DEFENSE INFORMATION SYSTEMS AGENCY
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IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **23 Sep 10**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Special Interoperability Test Certification of the Cisco Unified Communication Manager Local Session Controller, Version 8.0(2)

References: (a) DoD Directive 4630.05, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01E, "Interoperability and Supportability of Information Technology and National Security Systems," 15 December 2008
(c) through (e), see Enclosure 1

1. References (a) and (b) establish the Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The Cisco Unified Communications Manager (CUCM), Version 8.0(2), hereinafter referred to as the System Under Test (SUT) is certified for joint use in the Defense Information System Network (DISN) as a Local Session Controller (LSC). The Defense Information Systems Agency adjudicated all open non-AS Test Discrepancy Reports (TDRs) to have a minor operational impact. The fielding of the SUT is limited to IP version 4 (IPv4) across the DISN. Intra-enclave use of IPv4 and IPv6 is authorized for use. The certification status of the SUT will be verified during operational deployment. Any new discrepancy noted in the operational environment will be evaluated for impact on the existing certification. These discrepancies will be adjudicated to the satisfaction of DISA via a vendor PoAM, which will address all new critical TDRs within 120 days of identification. Testing was conducted using LSC product requirements derived from the Unified Capabilities Requirements (UCR), Reference (c), and LSC test procedures, Reference (d). No other configurations, features, or functions, except those cited within this memorandum, are certified by JITC. This certification expires upon changes that affect interoperability, but no later than three years from the date of this memorandum.

3. This finding is based on interoperability testing conducted by JITC, review of the vendor's Letters of Compliance (LoC), and DISA Information Assurance (IA) Certification Authority (CA) approval of the IA configuration. Interoperability testing was conducted by JITC, Fort Huachuca, Arizona, from 28 June 2010 through 20 August 2010. Review of the vendor's LoC was completed on 21 September 2010. The DISA CA has reviewed the IA Assessment Report for the SUT, Reference (e), and based on the findings in the report has provided a positive recommendation. The acquiring agency or site will be responsible for the DoD Information Assurance Certification and Accreditation Process (DIACAP) accreditation. The JITC certifies the SUT as meeting the UCR for LSC requirements. Enclosure 2 documents the test results and describes the tested network and system configurations including specified patch releases.

4. The interface, Capability Requirements (CR) and Functional Requirements (FR), and component status of the SUT is listed in Tables 1 and 2. The threshold Capability/Functional requirements for LSCs are established by Sections 5.3.2, 5.3.4, 5.3.5, and 5.4 of Reference (c) and were used to evaluate the interoperability of the SUT.

Table 1. SUT Interface Interoperability Status

Interface	Critical	UCR Reference	Threshold CR/FR Requirements (See note 1.)	Status	Remarks (See note 2.)
Line Interfaces					
10Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to PEIs and softphones.
100Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to PEIs and softphones.
1000Base-X	No	5.3.2.6.3	2, 4, 10,13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3ab. Applies to PEIs and softphones.
2-wire analog	Yes	5.3.2.6.1.6	2, 4, 10, and 13	Certified	Met threshold CRs/FRs for 2-wire instruments. Applies to 2-wire secure and non-secure analog instruments. Requirement met through use of an IAD integrated in the MG that supports IEEE 802.3i, 802.3u, and 802.3ab (See note 3.).
BRI	No	5.3.2.6.1.8	2, 4, 10, and 13	Not Tested	This interface is offered by the SUT but was not tested because it does not support Assured Services.
External Interfaces					
10Base-X	No (See note 4.)	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to AS-SIP trunk.
100Base-X	No (See note 4.)	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to AS-SIP trunk.
1000Base-X	No (See note 4.)	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3z and 802.3ab. Applies to AS-SIP trunk.
ISDN T1 PRI ANSI T1.619a	Yes	5.3.2.4.3	2, 3, 7, 8, 10, and 13	Certified	Met threshold CRs/FRs . Provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2	Yes	5.3.2.4.3	2, 3, 7, 8, 10, and 13	Certified	Met threshold CRs/FRs. Provides PSTN Connectivity
T1 CCS7 ANSI T1.619a	No	5.3.2.12.9	2 , 3, 7, 8, 10, and 13	Not Tested	This interface is not offered by the SUT.
T1 CAS	No	5.3.2.12.11	2, 3, 7, 8, 10, and 13	Not Certified	This interface is offered by the SUT but was not certified because of known discrepancies (See note 5.).
E1 PRI ITU-T Q.955.3	No (See note 6.)	5.3.2.12.10	2, 3, 7, 8, 10, and 13	Certified	Tested under PBX1 configuration. Results applicable to LSC.
E1 PRI ITU-T Q.931	No (See note 6.)	5.3.2.12.10	2, 3, 7, 8, 10, and 13	Certified	Conditionally required for European PSTN connectivity.

Table 1. SUT Interface Interoperability Status (continued)

Interface	Critical	UCR Reference	Threshold CR/FR Requirements (See note 1.)	Status	Remarks (See note 2.)
NM					
10Base-X	No (See note 4.)	5.3.2.4.4 5.3.2.7.2.8	16 and 17	Certified	Met threshold CRs/FRs. Verified via LoC.
100Base-X	No (See note 4.)	5.3.2.4.4 5.3.2.7.2.8	16 and 17	Certified	Met threshold CRs/FRs. Verified via LoC.

NOTES:
1. CR/FR requirements are contained in Table 2. CR/FR numbers represent a roll-up of UCR requirements. Enclosure 3 provides a list of more detailed requirements LSC products.
2. Paragraph 11 of Enclosure 2 provides detailed information pertaining to open TDRs and associated operational impacts
3. Voice calls from the SUT gateway analog interfaces via the UC DISN WAN require a loopback configuration of ANSI T1.619a ISDN PRI interfaces within each gateway (refer to Cisco CUCM deployment guide). This configuration requires translations in the gateways to route all out going analog calls placed towards the UC DISN WAN via the looped T1s. Additionally, incoming calls from the UC DISN WAN to analog end instruments on each gateway must be routed via the looped T1s. Without this configuration, analog end instruments cannot place calls via the UC DISN WAN. This configuration requires two looped ISDN PRI ANSI T1.619a T1s within each 3845 and 3945 gateways and will support a maximum of 69 analog interfaces per gateway. This allows for up to two ISDN PRI T1 interfaces or one ISDN PRI E1 interface for timing/network access. In addition, each 2851 and 2951 gateway requires one looped ANSI T1.619a ISDN PRI within each 2851 and 2951 gateway and will support a maximum of 23 analog interfaces per gateway. Both gateways also require a T1 or E1 interface for synchronization via recovered timing.
4. Must provide a minimum of one of the listed interfaces.
5. The SUT CAS interface had interoperability test discrepancies adjudicated to be critical for certification of this interface.
6. The interface is conditionally required for deployment in Europe.

LEGEND:

10Base-X	10 Mbps Ethernet
100Base-X	100 Mbps Ethernet
1000Base-X	1000 Mbps Ethernet
802.3ab	1000 Mbps Ethernet over Twisted Pair
802.3i	10 Mbps twisted pair media for 10Base-X networks
802.3j	10 Mbps fiber media for 10Base-X networks
802.3u	100BASE-TX, 100BASE-T4, 100BASE-FX Fast Ethernet at 100 Mbps with auto negotiation
802.3z	Standard for Gigabit Ethernet
ANSI	American National Standards Institute
AS-SIP	Assured Services Session Initiation Protocol
BRI	Basic Rate Interface
CAS	Channel Associated Signaling
CCS7	Common Channel Signaling
CR	Capability Requirement
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
FR	Functional Requirement
IAD	Integrated Access device
IEEE	Institute of Electrical and Electronics Engineers, Inc.
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector
LoC	Letter of Compliance
LSC	Local Session Controller
Mbps	Megabits per second
MG	Media Gateway
MLPP	Multi-Level Precedence and Preemption
NA	Not Applicable
NI-2	National ISDN Standard 2
NM	Network Management
PBX	Private Branch Exchange
PEI	Proprietary End Instrument
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.931	Signaling Standard for ISDN
Q.955.3	ISDN Signaling Standard for E1 MLPP
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
TDRs	Test Discrepancy Reports
UCR	Unified Capabilities Requirements
VoIP	Voice over Internet Protocol
WAN	Wide Area Network

Table 2. SUT Capability Requirements and Functional Requirements Status

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Status	Remarks
1	**Assured Services Product Features and Capabilities**				
	DSCP Packet Marking	Required	5.3.2.2.1.4	Met	
	Voice Features and Capabilities	Required	5.3.2.2.2.1	Partially Met	See note 2.
	Public Safety Features	Required	5.3.2.2.2.2	Met	
	ASAC – Open Loop	Required	5.3.2.2.2.3	Met	
	Signaling Protocols	Required	5.3.2.2.2.3	Met	
	Signaling Performance	Required	5.3.2.2.2.4	Met	
2	**Registration, Authentication, and Failover**				
	Registration	Required	5.3.2.3.1	Met	
	Failover	Required	5.3.2.3.2	Met	
3	**Product Physical, Quality, and Environmental Factors**				
	Availability	Required	5.3.2.5.2.1	Partially Met	See note 3
	Maximum Downtimes	Required	5.3.2.5.2.2	Met	
	Loss of Packets	Required (See note 4.)	5.3.2.5.4	Met	
4	**Voice End Instruments**				
	Tones and Announcements	Required	5.3.2.6.1.1	Partially Met	See notes 2 and 5.
	Audio Codecs	Required	5.3.2.6.1.2	Partially Met	See note 5.
	VoIP PEI or AEI Audio Performance Requirements	Required	5.3.2.6.1.3	Partially Met	See note 5.
	VoIP Sampling Standard	Required	5.3.2.6.1.4	Partially Met	See note 5.
	Authentication to LSC	Required	5.3.2.6.1.5	Partially Met	See note 5.
	Analog Telephone Support	Required (See note 6.)	5.3.2.6.1.6	Partially Met	See note 7.
	Softphones	Conditional	5.3.2.6.1.7	Met	See note 8.
	ISDN BRI	Conditional	5.3.2.6.1.8	Not Tested	
5	**Video End Instruments**				
	Video End Instrument	Required	5.3.2.6.2	Not Tested	See note 8.
	Display Messages, Tones, and Announcements	Required	5.3.2.6.2.1	Not Tested	See note 8.
	Video Codecs (Including Associated Audio Codecs)	Required	5.3.2.6.2.2	Not Tested	See note 8.
6	**LSC Requirements**				
	PBAS/ASAC Requirements	Required	5.3.2.7.2.1	Met	
	Calling Number Delivery Requirements	Required	5.3.2.7.2.2	Met	
	LSC Signaling Requirements	Required	5.3.2.7.2.3	Met	
	Service Requirements under Total Loss of WAN Transport	Required	5.3.2.7.2.4	Met	
	Local Location Server and Directory	Required	5.3.2.7.2.5	Met	
	LSC Transport Interface Functions	Required	5.3.2.7.2.7	Met	
	LSC to PEI, AEI, and Operator Console Status Verification	Required	5.3.2.7.2.10	Partially Met.	See note 9.
	Line-Side Custom Features Interference	Conditional	5.3.2.7.2.11	Met	
	Loop Avoidance	Required (See note 4.)	5.3.2.7.3	Met	

Table 2. SUT Capability Requirements and Functional Requirements Status (continued)

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Status	Remarks
7	**Call Connection Agent Requirements**				
	CCA IWF Component	Required (See note 10.)	5.3.2.9.2.1	Met	See note 11.
	CCA MGC Component	Required (See note 10.)	5.3.2.9.2.2	Met	
	SG Component	Conditional	5.3.2.9.2.3	Not Tested	See note 11.
	CCA-IWF Support for AS-SIP	Required	5.3.2.9.5.1	Met	
	CCA-IWF Support for SS7	Conditional	5.3.2.9.5.2	Not Tested	See note 11.
	CCA-IWF Support for PRI via MG	Required	5.3.2.9.5.3	Met	
	CCA-IWF Support for CAS Trunks via MG	Conditional	5.3.2.9.5.4	Not Tested	See note 11.
	CCA-IWF Support for PEI and AEI Signaling Protocols	Required	5.3.2.9.5.5	Partially Met	See note 12.
	CCA-IWF Support for VoIP and TDM Protocol Interworking	Required (See note 10.)	5.3.2.9.5.6	Met	See note 11.
	CCA Preservation of Call Ringing State during Failure Conditions	Required (See note 4.)	5.3.2.9.6	Met	
	CCA Interactions with Transport Interface Functions	Required	5.3.2.10.3	Met	
	CCA Interactions with the EBC	Required	5.3.2.10.4	Met	
	CCA Support for Admission Control	Required	5.3.2.10.5	Met	
	CCA Support for UFS	Required	5.3.2.10.6	Met	
	CCA Support for IA	Required	5.3.2.10.7	Met	
	CCA Interaction with EIs	Required	5.3.2.10.10	Partially Met	See note 5.
	CCA Support for AS Voice and Video	Required	5.3.2.10.11	Partially Met	See note 8.
	CCA Interactions with Service control Functions	Required	5.3.2.10.12	Met	
	CCA Interworking between AS-SIP and SS7	Conditional	5.3.2.11	Not Tested	See note 11.
8	**MG Requirements**				
	Role of MG In LSC	Required	5.3.2.12.3.1	Met	
	MG Support for ASAC	Required	5.3.2.12.4.1	Met	
	MG and IA Functions	Required	5.3.2.12.4.2	Met	
	MG Interaction with Service Control Function	Required	5.3.2.12.4.3	Met	
	MG Interactions with IP Transport Interface Functions	Required	5.3.2.12.4.4	Met	
	MG-EBC interactions	Required	5.3.2.12.4.5	Met	
	MG IP-Based PSTN Interface Requirements	Conditional	5.3.2.12.4.7	Not Tested	
	MG Interaction with EIs	Required	5.3.2.12.4.8	Met	See note 5.
	MG support for User Features and Services	Required	5.3.2.12.4.9	Met	
	MG Interface to TDM	Required	5.3.2.12.5	Met	See notes 10 and 11.
	MG Interface to TDM Allied and Coalition	Conditional	5.3.2.12.6	Not Tested	
	MG Interface to TDM PSTN in US	Required	5.3.2.12.7	Met	See note 10.
	MG Interfaces to TDM PSTN OCONUS	Required	5.3.2.12.8	Partially Met	See note 10.
	MG Support for CCS7	Conditional	5.3.2.12.9	Not Tested	See note 11.
	MG Support for ISDN PRI Trunks	Required	5.3.2.12.10	Met	

Table 2. SUT Capability Requirements and Functional Requirements Status (continued)

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Status	Remarks
8	**MG Requirements (continued)**				
	MG Support for CAS Trunks	Required	5.3.2.12.11	Not Tested	See note 11.
	MG requirements for VoIP Internal Interfaces	Required	5.3.2.12.12	Met	
	MG Echo Cancellation	Required	5.3.2.12.13	Met	
	MG Clock Timing	Required	5.3.2.12.14	Met	
	MGC-MG CCA Functions	Required	5.3.2.12.15	Met	
	MG V.150.1	Required	5.3.2.12.16	Not tested	See note 7.
	MG Preservation of Call Ringing during Failure	Required (See note 4.)	5.3.2.12.17	Met	
9	**SG Requirements**				
	SG and CCS7 Network Interactions	Conditional	5.3.2.13.5.1	Not Tested	See note 11.
	SG Interactions with CCA	Conditional	5.3.2.13.5.2	Not Tested	See note 11.
	SG Interworking Functions	Conditional	5.3.2.13.5.3	Not Tested	See note 11.
10	**WWNDP Requirements**				
	WWNDP	Required	5.3.2.16	Met	
	DSN WWNDP	Required	5.3.2.16.1	Met	
11	**Commercial Cost Avoidance**				
	Commercial Cost Avoidance	Required (See note 3.)	5.3.2.23	Not Tested	
12	**AS-SIP Based for External Devices (Voicemail, Unified Messaging, and Automated Receiving Devices)**				
	AS-SIP Requirements for External Interfaces	Conditional	5.3.2.24	Not Tested	
13	**Precedence Call Diversion**				
	Precedence call Diversion	Required	5.3.2.25	Met	
14	**Attendant Station Features**				
	Precedence and Preemption	Required (See note 3.)	5.3.2.26.1	Not Tested	See note 13.
	Call Display	Required (See note 3.)	5.3.2.26.2	Not Tested	See note 13.
	Class of Service Override	Required (See note 3.)	5.3.2.26.3	Not Tested	See note 13.
	Busy Override and Busy Verification	Required (See note 3.)	5.3.2.26.4	Not Tested	See note 13.
	Night service	Required (See note 3.)	5.3.2.26.5	Not Tested	See note 13.
	Automatic Recall of Attendant	Required (See note 3.)	5.3.2.26.6	Not Tested	See note 13
	Calls in Queue to the Attendant	Required (See note 3.)	5.3.2.26.7	Not Tested	See note 13.
15	**AS-SIP Requirements**				
	SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs	Required (See note 3.)	5.3.4.7	Not Tested	
	SIP Session Keep-Alive Timer	Required	5.3.4.8	Met	
	Session Description Protocol	Required	5.3.4.9	Met	
	Precedence and Preemption	Required	5.3.4.10	Met	
	Video Telephony – General Rules	Required	5.3.4.12	Not Tested	See note 8.
	Calling Services	Required	5.3.4.13	Met	
	SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances	Required	5.3.4.14	Met	
	Relevant Timers for the Terminating Gateway and the Originating Gateway	Required	5.3.4.15	Met	

Table 2. SUT Capability Requirements and Functional Requirements Status (continued)

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Status	Remarks
15	**AS-SIP Requirements (continued)**				
	SIP Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.16	Met	
	Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.17	Met	
	Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances	Required	5.3.4.18	Met	
	Supplementary Services	Required	5.3.4.19	Met	
16	**IPv6 Requirements**				
	Product Requirements	Required	5.3.5.4	Partially met	See note 14.
17	**NM**				
	LSC Management Function	Required	5.3.2.7.2.6	Met	
	VVoIP NMS Interface Requirements	Required	5.3.2.4.4	Met	
	General Management requirements	Required	5.3.2.17.2	Partially Met	See note 15.
	Requirement for FCAPS Management	Required	5.3.2.17.3	Partially Met	See note 15.
	NM requirements of Appliance Functions	Required	5.3.2.18	Partially Met	See note 15.
	Accounting Management	Required	5.3.2.19	Met	See note 16.

NOTES:
1. Annotation of 'required' refers to high level requirement category. Applicability of each sub-requirement is provided in enclosure 3.
2. The SUT had outstanding open TDRs at the completion of testing adjudicated by DISA to have a minor operational impact. The vendor has submitted a PoAM to address the open TDRs. Paragraph 11 of Enclosure 2 provides additional details.
3. When the SUT fails from the primary processor to backup processor all active drop after approx 6-8 minutes. DISA adjudicated this TDR as minor with the vendor's submitted PoAM to fix by June 2011.
4. This requirement represents a new UCR requirement where the vendor has 18-months (July 2011) to comply.
5. SUT met the requirement for PEIs; SUT was not tested with generic AEI because no AEI was provided. AEIs are a new UCR 2008 Change 1 requirement; the vendor has 18-months (July 2011) to comply.
6. UCR 2008 Change 1 added 18-month rule for G.711 and V.150.1 IAD support.
7. Vendor submitted LoC stating compliance to V.150 however this feature could not be tested because it is not supported by other vendors. This is a new UCR 2008 change 1 requirement; therefore the vendor has until July 2011 to comply with this requirement.
8. SUT did not demonstrate video requirements (conditional for softphone). Vendor did not provide a PEI video capability. This was adjudicated by DISA to have a low operational impact because of the limited deployment of PEIs with video.
9. SUT partially met PEI requirements (no video). The AEI and Operator Console requirements were not tested; the 18-month rule for complying (July 2011) applies.
10. The SUT must meet T1 PRI (T1.619a and NI-2) IWF. The T1 CAS and T1 CCS7 are conditional.
11. The SUT met T1/E1 PRI IWF requirements. The T1 CAS is supported but not certified and T1 CCS7 is not supported by the SUT.
12. The SUT met PEI CCA-IWF requirements. The AEI CCA-IWF requirements were not tested. The 18-month rule applies to AEIs.
13. The Attendant Console requirements are new UCR requirements; 18-month rule applies.
14. The SUT submitted an IPv6 LoC with noted discrepancies. Open TDRs were adjudicated by DISA to have a minor operational impact with vendor submitted PoAM.
15. The SUT submitted an NM LoC with noted discrepancies. Open TDRs were adjudicated by DISA to have a minor operational impact with vendor submitted PoAM.
16. The SUT does not comply with the objective requirement for Record Format.

Table 2. SUT Capability Requirements and Functional Requirements Status (continued)

LEGEND:

AEI	AS-SIP End Instrument	LSC	Local Session Controller
AS	Assured Services	Mbps	Megabits per second
AS-SIP	Assured Services Session Initiation Protocol	MG	Media Gateway
BRI	Basic Rate Interface	MGC	Media Gateway Controller
C2	Command and Control	MFSS	Multi-Function Soft Switch
CAS	Channel Associated Signaling	MLPP	Multilevel Precedence and Preemption
CCA	Call Connection Agent	NI-2	National ISDN Standard 2
CR	Capabilities Requirement	NM	Network Management
CCS7	Common Channel Signaling	NMS	Network Management System
DHCP	Dynamic Host Configuration Protocol	OCONUS	Outside the Continental United States
DISA	Defense Information Systems Agency	PBAS	Precedence Based Assured Services
DSCP	Differentiated Services Code Point	PEI	Proprietary End Instrument
DSN	Defense Switched Network	PoAM	Plan of Action and Milestones
EBC	Edge Boarder Controller	PRI	Primary Rate Interface
EI	End Instrument	PSTN	Public Switched Telephone Network
FCAPS	Fault, Configuration, Accounting, Performance and Security	SG	Signaling Gateway
		SIP	Session Initiation Protocol
FR	Functional Requirement	SS7	Signaling System 7
G.711	Standard for PCM of Voice Frequencies	SUT	System Under Test
IA	Information Assurance	T1	Digital Transmission Link Level 1 (1.544 Mbps)
IAD	Integrated Access Device	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
IP	Internet Protocol	TDM	Time Division Multiplexing
ID	Identification	TDR	Test Discrepancy Report(s)
ISDN	Integrated Services Digital Network	UCR	Unified Capabilities Requirements
IEEE	Institute of Electrical and Electronics Engineers, Inc.	UFS	User Features and Services
IP	Internet Protocol	U.S.	United States
IPv6	Internet Protocol version 6	VoIP	Voice over Internet Protocol
IWF	Interworking Function	WAN	Wide Area Network
JITC	Joint Interoperability Test Command	WWNDP	Worldwide Numbering and Dialing Plan
LoC	Letter of Compliance		

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi. All associated data is available on the Defense Information Systems Agency Unified Capability Coordination Office (UCCO) website located at http://www.disa.mil/ucco/.

6. The JITC point of contact is Mr. Edward Mellon, commercial (520) 538-5159, or DSN 312-879-5159, e-mail address is edward.mellon@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number is 1011801.

FOR THE COMMANDER:

3 Enclosures a/s

for RICHARD A. MEADOR
Chief
Battlespace Communications Portfolio

Distribution (electronic mail):
Joint Staff J-6
Joint Interoperability Test Command, Liaison, TE3/JT1
Office of Chief of Naval Operations, CNO N6F2
Headquarters U.S. Air Force, Office of Warfighting Integration & CIO, AF/XCIN (A6N)
Department of the Army, Office of the Secretary of the Army, DA-OSA CIO/G-6 ASA (ALT), SAIS-IOQ
U.S. Marine Corps MARCORSYSCOM, SIAT, MJI Division I
DOT&E, Net-Centric Systems and Naval Warfare
U.S. Coast Guard, CG-64
Defense Intelligence Agency
National Security Agency, DT
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Office of Assistant Secretary of Defense (NII)/DOD CIO
U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities Division, J68
Defense Information Systems Agency, GS23

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ADDITIONAL REFERENCES

(c) Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008, Change 1," 22 January 2010
(d) Joint Interoperability Test Command, "Unified Capabilities Test Plan (UCTP),"
(e) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of Cisco Unified Communications Manager (CUCM), Version 8.02, (TN 1011801),"

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CERTIFICATION TESTING SUMMARY

1. SYSTEM TITLE. The Cisco Unified Communication Manager Local Session Controller (LSC), Version 8.0(2) with specified releases.

2. SPONSOR. United States Air Force, Attention: Joseph Halcli, HQ USAFE/A6NA, Address: PSC2 Box 11095, APO AE, 09012, Phone: 314-478-0520, e-mail: joseph.halcli@ramstein.af.mil.

3. PROGRAM MANAGER. Louis Schmuckler GS15, Voice Services Engineer Branch, GS241, PO Box 4502, Arlington VA, 22204-4502, e-mail: louis.schmuckler@disa.mil.

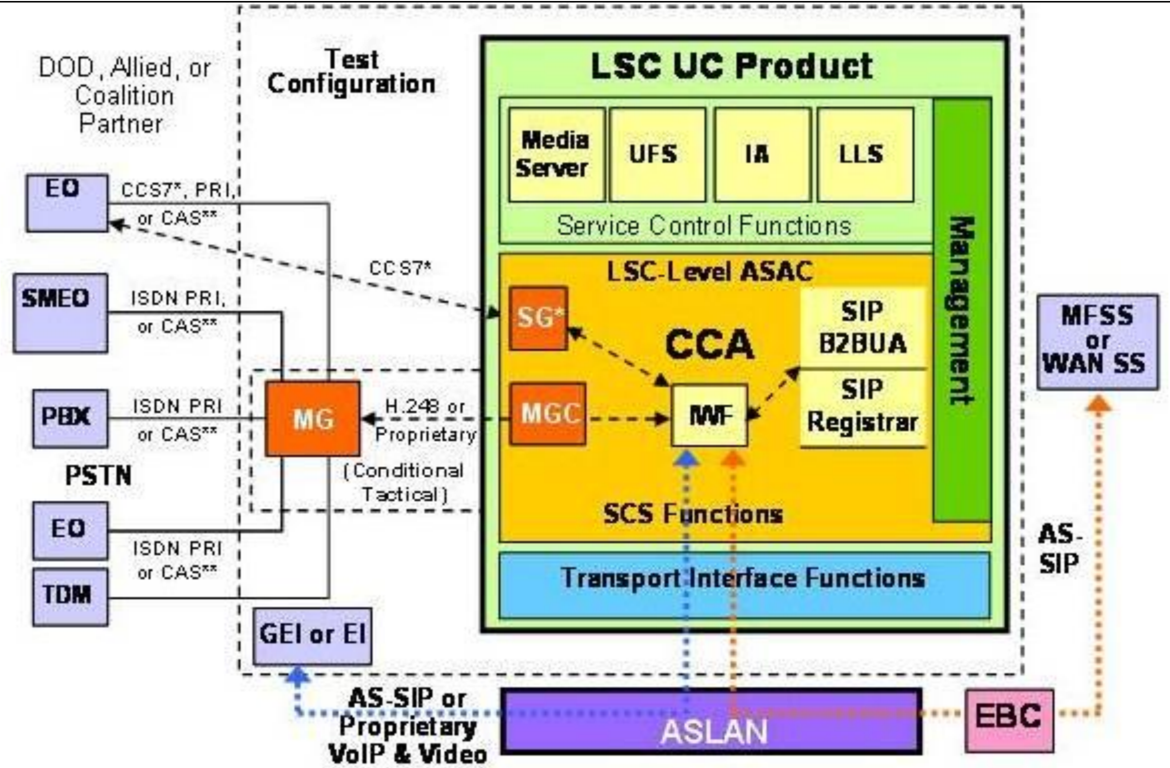
4. TESTER. Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.

5. SYSTEM DESCRIPTION. The SUT is an enterprise IP telephony call-processing solution that is scalable, and distributable. Multiple Cisco Unified Communications Manager servers are clustered and managed as a single entity on an IP network, a distinctive capability in the industry that yields scalability of 1 to 30,000 IP phones per cluster, load balancing, and call-processing service redundancy. The SUT is installed on the Cisco 7800 Series Media Convergence Servers (MCSs) platforms, the B200-M1 blade servers with VMware EXSi in the Unified Computing System 5108 chassis and/or the Unified Computing System C210 Rack Mount Servers with VMware ESXi. The SUT supports two types of user End Instruments connection types, IP Phones (both hardware and software based) and Analog devices connected to an Integrated Services Router (ISR – both first and second generation versions were tested). The Cisco 2851/2951 and 3845/3945 (ISR/ISR-G2s) are all also used as VoIP gateways with traditional TDM circuits such as T1/E1 trunks. The Gateway routers provide connectivity from VoIP networks to legacy Time Division Multiplexing (TDM) products. The SUT's software was tested on the following server platforms; MCS7835I3, MCS7835H2, MCS7825I4, UCS5108 with B200-M1 (with VMware) and UCS C210-M1. The SUT is optimized to run on select configurations of the Cisco Unified Computing System. With the Cisco Unified Computing System, applications run in a virtualized environment comprised of VMware software and Cisco Unified Computing System servers. Supported hardware configurations can be found by selecting the "Cisco Unified Communications on the Cisco Unified Computing System" link at the following URL: www.cisco.com/go/swonly. These configurations are certified by similarity to the systems that were tested at JITC. The other family series of servers which include: MCS7835I2, MCS7825H3, MCS7825H4, MCS7835H3, MCS7845H3, MCS7845H2, MCS7845I3, MCS7845I2 and B250-M1 utilize the same software and similar hardware respectively, and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use within the UC DISN. The SUT was tested with the 3845, 3945, 2951, and 2851 Integrated Services Routers (ISR). The other ISR in the family to include the 3825, 3945E, 3925, 3925E, 2921, 2821, 2911 and 2811 utilize the same software and similar hardware respectively, and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use within the UC DISN.

Voice calls from the SUT gateway analog interfaces via the UC DISN WAN require a unique configuration of ANSI T1.619a ISDN PRI interfaces within each gateway (refer to Cisco UCM deployment guide). This configuration requires translations in the gateways to route all out going analog calls placed towards the UC DISN WAN via the looped T1s. Additionally, incoming calls from the UC DISN WAN to analog end instruments on each gateway must be routed via these T1s. Without this configuration, analog end instruments cannot place calls via the UC DISN WAN. This configuration requires two looped ISDN PRI ANSI T1.619a T1s within each 3845 and 3945 gateways and will support a maximum of 69 analog interfaces per gateway. This allows for up to two ISDN PRI T1 interfaces or one ISDN PRI E1 interface for timing/network access. In addition, each 2851 and 2951 gateway requires one looped ANSI T1.619a ISDN PRI within each 2851 and 2951 gateway and will support a maximum of 23 analog interfaces per gateway. Both gateways also require a T1 or E1 interface for synchronization via recovered timing.

The SUT provides voice and video services, legacy 2-wire analog telephones, Internet Protocol (IP) telephones, and media processing devices within a local service domain. Since the SUT video end instruments can only support intra-enclave calls they are not certified for use within the UC DISN. The SUT offers an analog voice gateway (VG) 224 that services 24 analog users. This gateway does not support V.150.1 modem over IP and is only certified for non secure voice and facsimile. In accordance with UCR 2008 change 1 paragraph 5.3.2.6.1.6 every analog Integrated Access Device (IAD) line card on the LSC or MFSS is not required to support secure voice, secure data, or non secure modem or ITU–T Recommendation V.150.1 Modem Relay. The other SUT gateways support secure voice and data, and V.150.1 modem relay.

6. OPERATIONAL ARCHITECTURE. Figure 2-1 depicts the LSC functional model and Figure 2-2 the notional operational architecture that the SUT may be used in.

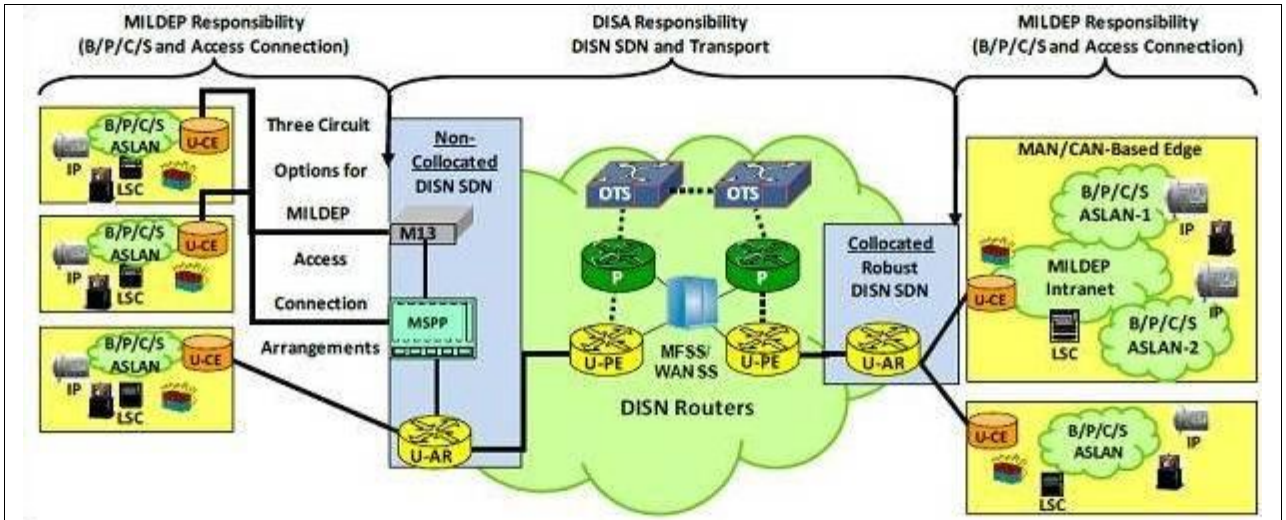


*Support for an SG & CCS7 is Conditional for an LSC.
**Support for CAS is Conditional for an LSC.
NOTE: The EI (using Proprietary VoIP & Video) is part of the LSC UC product, the GEI (AS-SIP VoIP & Video) is not.

LEGEND:

ASAC	Assured Service Admission Control	LSC	Local Session Controller
ASLAN	Assured Services Local Area Network	MFS	Multifunction Switch
AS-SIP	Assured Services Session Initiation Protocol	MFSS	Multifunction Softswitch
B2BUA	Back-to-Back User Agent	MG	Media Gateway
CAS	Channel Associated Signaling	MGC	Media Gateway Controller
CCA	Call Connection Agent	PBX	Private Branch Exchange
CCS7	Common Channel Signaling No. 7	PRI	Primary Rate Interface
DOD	Department of Defense	PSTN	Public Switched Telephone Network
EBC	Edge Border Controller	SCS	Session Control and Signaling
EI	End Instrument	SG	Signaling Gateway
EO	End Office	SIP	Session Initiation Protocol
GEI	Generic End Instrument	SS	Softswitch
IA	Information Assurance	TDM	Time Division Multiplexer
ISDN	Integrated Services Digital Network	UC	Unified Capabilities
IWF	Interworking Function	UFS	User Features and Services
LLS	Local Location Service	WAN	Wide Area Network
LSC	Local Session Controller	VoIP	Voice over Internet Protocol

Figure 2-1. LSC Functional Reference Model



LEGEND:

ASLAN	Assured Services Local Area Network	MILDEP	Military Department
B/P/C/S	Base, Post, Camp, Station	MSPP	Multi-Service Provisioning Platform
CAN	Campus Area Network	OTS	Optical Transport System
DISA	Defense Information Systems Agency	P	Provider
DISN	Defense Information System Network	SDN	Service Delivery Node
IP	Internet Protocol	SS	Softswitch
LSC	Local Session Controller	U-AR	Unclassified Aggregation Router
M13	Multiplexer	U-CE	Unclassified Customer Edge Router
MAN	Metropolitan Area Network	U-PE	Unclassified Provider Edge Router
MFSS	Multifunction Softswitch	WAN	Wide Area Network

Figure 2-2. UC Network Architecture

7. INTEROPERABILITY REQUIREMENTS. The interface, Capability Requirements (CR) and Functional Requirements (FR), Information Assurance (IA), and other requirements for LSCs are established by Sections 5.3.2, 5.3.4, 5.3.5, and 5.4 of Reference (c).

7.1 Interfaces. The CUCM 8.0(2) uses the external interfaces to connect to the Global Information Grid (GIG) network and other Unified Capabilities products. Table 2-1, shows the physical interfaces supported by the SUT. The table documents the physical interfaces and the associated standards.

Table 2-1. LSC Interface Requirements

Interface	Critical	UCR Reference	Criteria	Remarks
Line Interfaces				
10Base-X	Yes	5.3.2.6.3	Support minimum threshold CRs/FRs (2, 4,10, 13, and 16) and meet interface criteria for 802.3i and 802.3j	
100Base-X	Yes	5.3.2.6.3	Support minimum threshold CRs/FRs (2, 4,10, 13, and 16) and meet interface criteria for 802.3.u	

Table 2-1. LSC Interface Requirements (continued)

Interface	Critical	UCR Reference	Criteria (See note 1.)	Remarks
Line Interfaces (continued)				
1000Base-X	No	5.3.2.6.3	Support minimum threshold CRs/FRs (2, 4, 10, 13, and 16) and meet interface criteria for 802.3z.	
2-wire analog	Yes	5.3.2.6.1.6	Support minimum threshold CRs/FRs (2, 4, 10, and 13) and meet interface criteria for analog.	
BRI	No	5.3.2.6.1.8	Support minimum threshold CRs/FRs (2, 4, 10, and 13) and meet interface criteria for BRI	
External Interfaces				
10Base-X	No (See note 2.)	5.3.2.4.2	Support minimum threshold CRs/FRs (1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16) and meet interface criteria for 802.3i and 802.3j	
100Base-X	No (See note 2.)	5.3.2.4.2	Support minimum threshold CRs/FRs (1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16) and meet interface criteria for 802.3u	
1000Base-X	No (See note 2.)	5.3.2.4.2	Support minimum threshold CRs/FRs (1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16)and meet interface criteria for 802.3z	
ISDN T1 PRI ANSI T1.619a	Yes	5.3.2.4.3	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for ISDN T1 PRI (T1.619a)	Provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2	Yes	5.3.2.4.3	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13)and meet interface criteria for ISDN T1 PRI (NI-2)	Provides PSTN Connectivity.
T1 CCS7 ANSI T1.619a	No	5.3.2.12.9	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for T1 CCS7 (ANSI T1.619a)	
T1 CAS	No	5.3.2.12.11	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for T1 CAS	T1 CAS with MLPP.
E1 PRI ITU-T Q.955.3	No	5.3.2.12.10	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13)and meet interface criteria for EI PRI (Q.955.3)	Conditionally required for DSN European connectivity.
E1 PRI ITU-T Q.931	No	5.3.2.12.10	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13)and meet interface criteria for E1 PRI (ITU-T Q.931)	Conditionally required for commercial European connectivity.
NM				
10Base-X	No (See note 2.)	5.3.2.4.4 5.3.2.7.2.8	Support minimum threshold CR/FRs (16 and 17) and meet interface criteria for 802.3i and 802.3j	
100Base-X	No (See note 2.)	5.3.2.4.4 5.3.2.7.2.8	Support minimum threshold CR/FRs (16 and 17) and meet interface criteria for 802.3u	

NOTES:
1. CR/FR requirements are contained in Table 2-2. CR/FR numbers represent a roll-up of UCR requirements. Enclosure 3 provides a list of more detailed requirements for security device products.
2. Must provide a minimum of one of the listed interfaces.

LEGEND:

ANSI	American National Standards Institute
BRI	Basic Rate Interface
CR	Capability Requirement
CCS7	Common Channel Signaling
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
FR	Functional Requirement
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector
LSC	Local Session Controller
Mbps	Megabits per second
MLPP	Multi-Level Precedence and Preemption
NI-2	National ISDN Standard 2
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
T1	Digital Transmission Link Level 1 (1.544 Mbps)
UCR	Unified Capabilities Requirements

7.2 Capability Requirements (CR) and Functional Requirements (FR). The LSCs have required and conditional features and capabilities that are established by Sections 5.3.2, 5.3.4, 5.3.5, and 5.4 of the UCR. The SUT does not need to provide non-critical (conditional) requirements. If they are provided, they must function according to the specified requirements. The SUTs features and capabilities and its aggregated requirements are listed in Table 2-2. Detailed CR/FR requirements are provided in Table 3-1 of Enclosure 3.

Table 2-2. LSC Capability Requirements and Functional Requirements

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Criteria
1	**Assured Services Product Features and Capabilities**			
	DSCP Packet Marking	Required	5.3.2.2.1.4	See note 2.
	Voice Features and Capabilities	Required	5.3.2.2.2.1	
	Public Safety Features	Required	5.3.2.2.2.2	
	ASAC – Open Loop	Required	5.3.2.2.3	
	Signaling Protocols	Required	5.3.2.2.2.3	
	Signaling Performance	Required	5.3.2.2.4	
2	**Registration, Authentication, and Failover**			
	Registration	Required	5.3.2.3.1	See note 2.
	Failover	Required	5.3.2.3.2	
3	**Product Physical, Quality, and Environmental Factors**			
	Availability	Required	5.3.2.5.2.1	See note 2.
	Maximum Downtimes	Required	5.3.2.5.2.2	
	Loss of Packets	Required (See note 3.)	5.3.2.5.4	
4	**Voice End Instruments**			
	Tones and Announcements	Required	5.3.2.6.1.1	See note 2.
	Audio Codecs	Required	5.3.2.6.1.2	
	VoIP PEI or AEI Audio Performance Requirements	Required	5.3.2.6.1.3	
	VoIP Sampling Standard	Required	5.3.2.6.1.4	
	Authentication To LSC	Required	5.3.2.6.1.5	
	Analog Telephone Support	Required (See note 4.)	5.3.2.6.1.6	
	Softphones	Conditional	5.3.2.6.1.7	
	ISDN BRI	Conditional	5.3.2.6.1.8	
5	**Video End Instruments**			
	Video End Instrument	Required	5.3.2.6.2	See note 2.
	Display Messages, Tones, and Announcements	Required	5.3.2.6.2.1	
	Video Codecs (Including Associated Audio Codecs)	Required	5.3.2.6.2.2	
6	**LSC Requirements**			
	PBAS/ASAC Requirements	Required	5.3.2.7.2.1	See note 2.
	Calling Number Delivery Requirements	Required	5.3.2.7.2.2	
	LSC Signaling Requirements	Required	5.3.2.7.2.3	
	Service Requirements under Total Loss of WAN Transport	Required	5.3.2.7.2.4	
	Local Location Server and Directory	Required	5.3.2.7.2.5	
	LSC Transport Interface Functions	Required	5.3.2.7.2.7	
	LSC to PEI, AEI, and Operator Console Status Verification	Required	5.3.2.7.2.10	
	Line-Side Custom Features Interference	Conditional	5.3.2.7.2.11	
	Loop Avoidance	Required (See note 3.)	5.3.2.7.3	

Table 2-2. LSC Capability Requirements and Functional Requirements (continued)

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Criteria
7	**Call Connection Agent Requirements**			
	CCA IWF Component	Required (See note 5.)	5.3.2.9.2.1	See note 2.
	CCA MGC Component	Required (See note 5.)	5.3.2.9.2.2	
	SG Component	Conditional	5.3.2.9.2.3	
	CCA-IWF Support for AS-SIP	Required	5.3.2.9.5.1	
	CCA-IWF Support for SS7	Conditional	5.3.2.9.5.2	
	CCA-IWF Support for PRI via MG	Required	5.3.2.9.5.3	
	CCA-IWF Support for CAS Trunks via MG	Conditional	5.3.2.9.5.4	
	CCA-IWF Support for PEI and AEI Signaling Protocols	Required	5.3.2.9.5.5	
	CCA-IWF Support for VoIP and TDM Protocol Interworking	Required (See note 5.)	5.3.2.9.5.6	
	CCA Preservation of Call Ringing State during Failure Conditions	Required	5.3.2.9.6	
	CCA Interactions with Transport Interface Functions	Required	5.3.2.10.3	
	CCA Interactions with the EBC	Required	5.3.2.10.4	
	CCA Support for Admission Control	Required	5.3.2.10.5	
	CCA Support for UFS	Required	5.3.2.10.6	
	CCA Support for IA	Required	5.3.2.10.7	
	CCA Interaction with EIs	Required	5.3.2.10.10	
	CCA Support for AS Voice and Video	Required	5.3.2.10.11	
	CCA Interactions with Service control Functions	Required	5.3.2.10.12	
	CCA Interworking between AS-SIP and SS7	Conditional	5.3.2.11	
8	**MG Requirements**			
	Role of MG In LSC	Required	5.3.2.12.3.1	See note 2.
	MG Support for ASAC	Required	5.3.2.12.4.1	
	MG and IA Functions	Required	5.3.2.12.4.2	
	MG Interaction with Service Control Function	Required	5.3.2.12.4.3	
	MG Interactions with IP Transport Interface Functions	Required	5.3.2.12.4.4	
	MG-EBC interactions	Required	5.3.2.12.4.5	
	MG IP-Based PSTN Interface Requirements	Conditional	5.3.2.12.4.7	
	MG Interaction with EIs	Required	5.3.2.12.4.8	
	MG support for User Features and Services	Required	5.3.2.12.4.9	
	MG Interface to TDM	Required (See note 6.)	5.3.2.12.5	
	MG Interface to TDM Allied and Coalition	Conditional	5.3.2.12.6	
	MG Interface to TDM PSTN in US	Required (See note 6.)	5.3.2.12.7	
	MG Interfaces to TDM PSTN OCONUS	Required (See notes 6 and 7.)	5.3.2.12.8	
	MG Support for CCS7	Conditional	5.3.2.12.9	

Table 2-2. LSC Capability Requirements and Functional Requirements (continued)

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Criteria
8	**MG Requirements (continued)**			
	MG Support for ISDN PRI Trunks	Required	5.3.2.12.10	See note 2.
	MG Support for CAS Trunks	Required	5.3.2.12.11	
	MG requirements for VoIP Internal Interfaces	Required	5.3.2.12.12	
	MG Echo Cancellation	Required	5.3.2.12.13	
	MG Clock Timing	Required	5.3.2.12.14	
	MGC-MG CCA Functions	Required	5.3.2.12.15	
	MG V.150.1	Required	5.3.2.12.16	
	MG Preservation of Call Ringing during Failure	Required (See note 3.)	5.3.2.12.17	
9	**SG Requirements**			
	SG and CCS7 network Interactions	Conditional	5.3.2.13.5.1	See note 2.
	SG Interactions with CCA	Conditional	5.3.2.13.5.2	
	SG Interworking Functions	Conditional	5.3.2.13.5.3	
10	**WWNDP Requirements**			
	WWNDP	Required	5.3.2.16	See note 2.
	DSN WWNDP	Required	5.3.2.16.1	
11	**Commercial Cost Avoidance**			
	Commercial Cost Avoidance	Required (See note 3.)	5.3.2.23	See note 2.
12	**AS-SIP Based for External Devices (Voicemail, Unified Messaging, and Automated Receiving Devices)**			
	AS-SIP Requirements for External Interfaces	Conditional	5.3.2.24	See note 2.
13	**Precedence Call Diversion**			
	Precedence call Diversion	Required	5.3.2.25	See note 2.
14	**Attendant Station Features**			
	Precedence and Preemption	Required (See note 3.)	5.3.2.26.1	See note 2.
	Call Display	Required (See note 3.)	5.3.2.26.2	
	Class of Service Override	Required (See note 3.)	5.3.2.26.3	
	Busy Override and Busy Verification	Required (See note 3.)	5.3.2.26.4	
	Night service	Required (See note 3.)	5.3.2.26.5	
	Automatic Recall of Attendant	Required (See note 3.)	5.3.2.26.6	
	Calls in Queue to the Attendant	Required (See note 3.)	5.3.2.26.7	
15	**AS-SIP Requirements**			
	SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs	Required (See note 3.)	5.3.4.7	See note 2.
	SIP Session Keep-Alive Timer	Required	5.3.4.8	
	Session Description Protocol	Required	5.3.4.9	
	Precedence and Preemption	Required	5.3.4.10	
	Video Telephony – General Rules	Required	5.3.4.12	
	Calling Services	Required	5.3.4.13	

Table 2-2. LSC Capability Requirements and Functional Requirements (continued)

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Criteria
15	**AS-SIP Requirements (continued)**			
	SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances	Required	5.3.4.14	See note 2.
	Relevant Timers for the Terminating Gateway and the Originating Gateway	Required	5.3.4.15	
	SIP Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.16	
	Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.17	
	Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances	Required	5.3.4.18	
	Supplementary Services	Required	5.3.4.19	
16	**IPv6 Requirements**			
	Product Requirements	Required	5.3.5.4	See note 2.
17	**NM**			
	LSC Management Function	Required	5.3.2.7.2.6	See note 2.
	VVoIP NMS Interface Requirements	Required	5.3.2.4.4	
	General Management requirements	Required	5.3.2.17.2	
	Requirement for FCAPS Management	Required	5.3.2.17.3	
	NM requirements of Appliance Functions	Required	5.3.2.18	
	Accounting Management	Required	5.3.2.19	

NOTES:

1. Annotation of ‘required’ refers to high level requirement category. Applicability of each sub-requirement is provided in Enclosure 3.
2. Detailed requirements and associated criteria for LSC s are listed in Table 3-1 of Enclosure 3.
3. This requirement represents a new UCR requirement for which the vendor has 18-months (July 2011) to comply.
4. The UCR 2008 Change 1 added 18-month rule for G.711 and V.150.1 IAD support.
5. The LSC must meet T1 PRI (ANSI T1.619a and NI-2) CCA IWF. The T1 CAS and T1 CCS7 CCA IWF are conditional.
6. The LSC must meet TDM requirements for T1 PRI (ANSI T1.619a and NI-2). The TDM requirements for T1 CAS and T1 CCS7 are conditional.
7. The E1 requirements for OCOUNUS are conditionally required for deployments in Europe.

Table 2-2. LSC Capability Requirements and Functional Requirements (continued)

LEGEND:

AEI	AS-SIP En Instrument	LoC	Letter of Compliance
AS	Assured Services	LSC	Local Session Controller
AS-SIP	Assured Services Session Initiation Protocol	Mbps	Megabits per second
BRI	Basic Rate Interface	MG	Media Gateway
C2	Command and Control	MGC	Media Gateway Controller
CAS	Channel Associated Signaling	MFSS	Multi-Function Soft Switch
CCA	Call Connection Agent	MLPP	Multilevel Precedence and Preemption
CR	Capabilities Requirement	NI-2	National ISDN Standard 2
CCS7	Common Channel Signaling	NM	Network Management
DHCP	Dynamic Host Configuration Protocol	NMS	Network Management System
DISA	Defense Information Systems Agency	OCONUS	Outside the Continental United States
DSCP	Differentiated Services Code Point	PBAS	Precedence Based Assured Services
DSN	Defense Switched Network	PEI	Proprietary End Instrument
EBC	Edge Boarder Controller	PoAM	Plan of Action and Milestones
EI	End Instrument	PRI	Primary Rate Interface
FCAPS	Fault, Configuration, Accounting, Performance and Security	PSTN	Public Switched Telephone Network
		SG	Signaling Gateway
FR	Functional Requirement	SIP	Session Initiation Protocol
G.711	Standard for PCM of Voice Frequencies	SS7	Signaling System 7
IA	Information Assurance	SUT	System Under Test
IAD	Integrated Access Device	T1	Digital Transmission Link Level 1 (1.544 Mbps)
IP	Internet Protocol	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
ID	Identification	TDM	Time Division Multiplexing
ISDN	Integrated Services Digital Network	TDR	Test Discrepancy Report(s)
IEEE	Institute of Electrical and Electronics Engineers, Inc.	UCR	Unified Capabilities Requirements
IP	Internet Protocol	UFS	User Features and Services
IPv6	Internet Protocol version 6	U.S.	United States
IWF	Interworking Function	VoIP	Voice over Internet Protocol
JITC	Joint Interoperability Test Command	WAN	Wide Area Network
		WWNDP	Worldwide Numbering and Dialing Plan

7.3 Information Assurance. Table 2-3 details the Information Assurance (IA) requirements applicable to an LSC.

Table 2-3. LSC IA Requirements

Requirement	Applicability (See note.)	UCR Reference	Criteria
General Requirements	Required	5.4.6.2	Detailed requirements and associated criteria for LSC are listed in the IATP (Reference (e)).
Authentication	Required	5.4.6.2.1	
Integrity	Required	5.4.6.2.2	
Confidentiality	Required	5.4.6.2.3	
Non-Repudiation	Required	5.4.6.2.4	
Availability	Required	5.4.6.2.5	

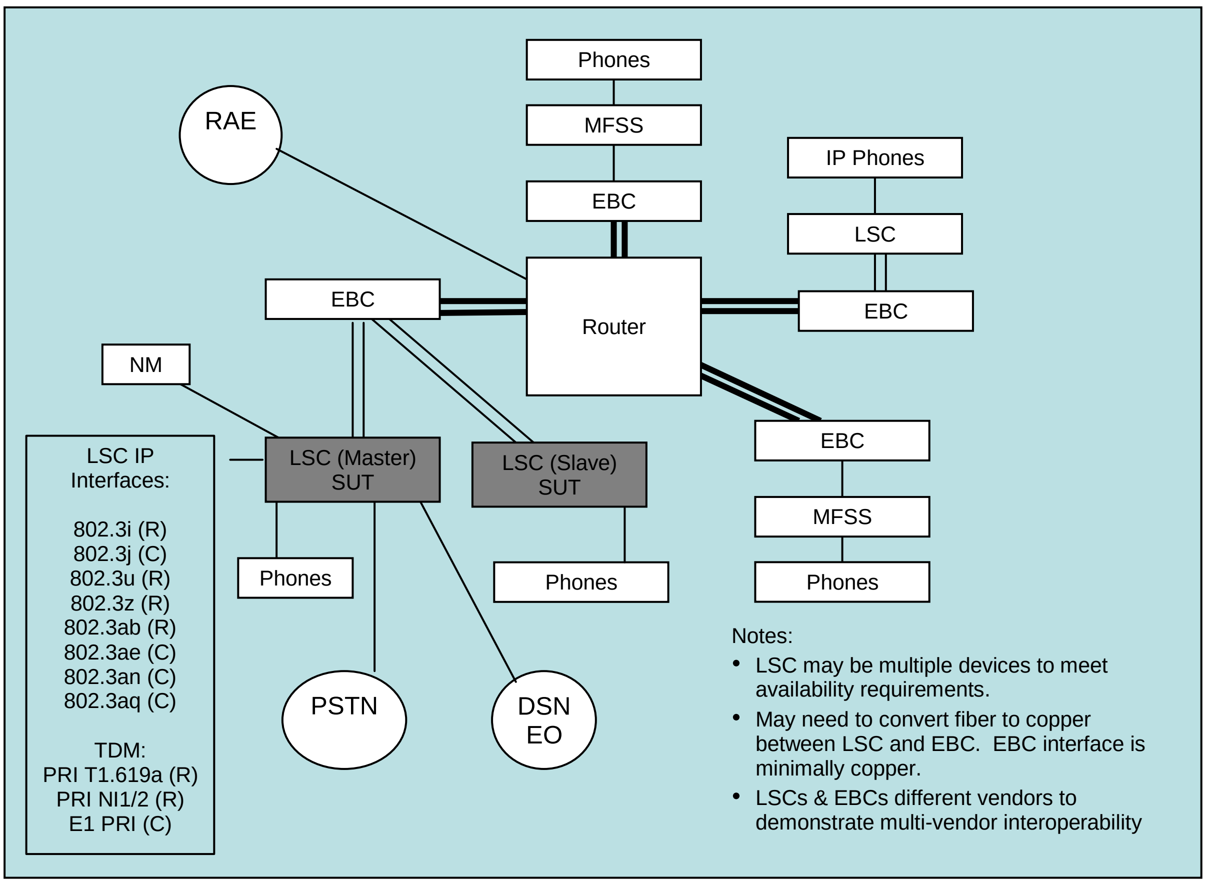
NOTE: Annotation of 'required' refers to high level requirement category. Applicability of each sub-requirement is provided in Reference (e).

LEGEND:

IA	Information Assurance	LSC	Local Session Controller
IATP	IA Test Plan	UCR	Unified capabilities Requirements

7.4 Other. None.

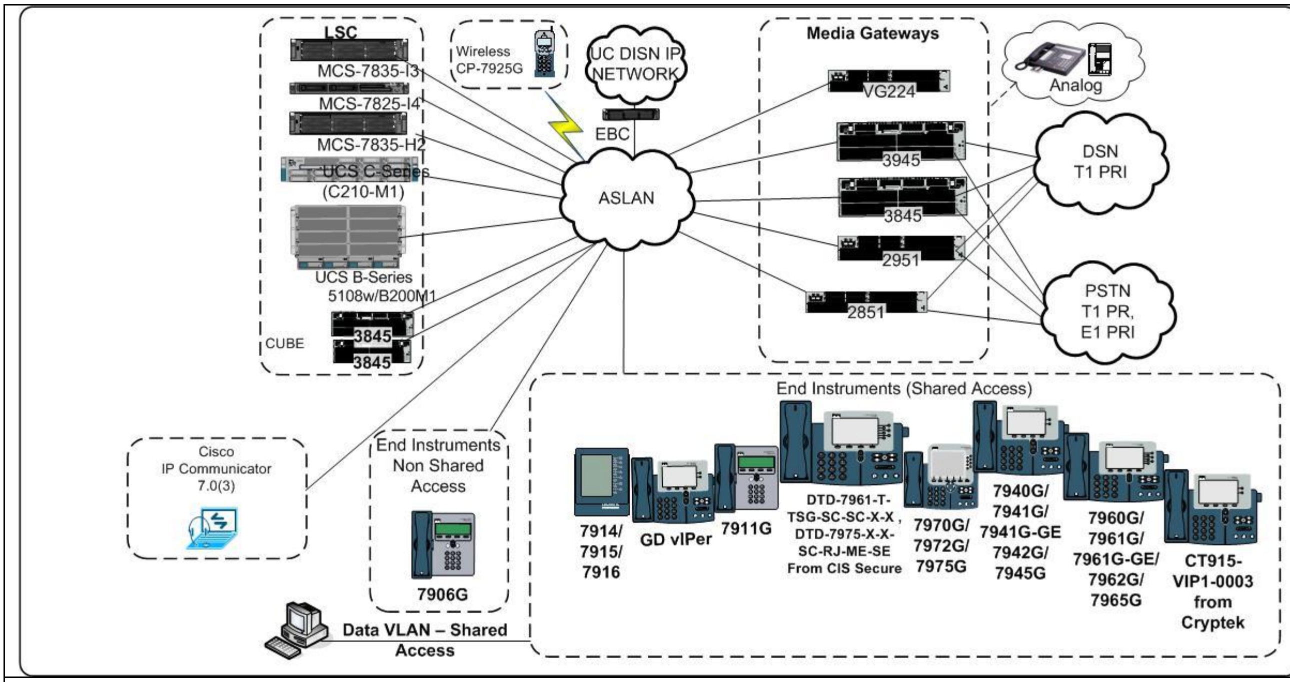
8. TEST NETWORK DESCRIPTION. The SUT was tested at Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona in a manner and configuration similar to that of a notional operational environment. Testing the system's required functions and features was conducted using the test configurations depicted in Figures 2-3 and 2-4. Figure 2-3 depicts the minimum test architecture for testing LSCs. Figure 2-4 depicts the SUT's test configuration.



LEGEND:

C	Conditional	NI 1/2	National ISDN Standard 1 or 2
DSN	Defense Switched Network	NM	Network Management
EBC	Edge Boundary Controller	PRI	Primary Rate Interface
E1	European Basic Multiplex Rate (2.048 Mbps)	PSTN	Public Switched Telephone Network
EO	End Office	R	Required
IP	Internet Protocol	RAE	Required Ancillary Equipment
ISDN	Integrated Services Digital Network	SUT	System Under Test
LSC	Local Session Controller	T1	Digital Transmission Link Level 1 (1.544 Mbps)
Mbps	Megabits per second	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
MFSS	Multifunction Softswitch	TDM	Time Division Multiplexing
MLPP	Multi-Level Precedence and Preemption		

Figure 2-3. LSC Minimum Test Architecture



Note: All components enclosed in dashed lines are the SUT.

LEGEND:

ASLAN	Assured Services Local Area Network
CAS	Channel Associated Signaling
DISN	Defense Information System Network
DSN	Defense Switching Network
E1	European Basic Multiplex Rate (2.048 Mbps)
EBC	Edge Boundary Controller
GD	General Dynamics
IP	Internet Protocol
LSC	Local Session Controller
Mbps	Megabits per second
MCS	Media Convergence Server
PRI	Primary Rate Interface
SUT	System Under Test
T1	Digital Transmission Link Level 1(1.544 Mbps)
UC	Unified Capabilities
VLAN	Virtual Local Area Network

Figure 2-4. SUT Test Configuration

9. SYSTEM CONFIGURATIONS. Table 2-4 provides the system configurations and hardware and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine its interoperability capability with associated network devices and network traffic.

Table 2-4. SUT Tested System Configurations

Cisco Unified Communications Manager Version 8.0(2), with IOS Software Release 15.1(1)T			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Function**
Communication Managers **MCS7835I3**, **MCS7835H2**, **MCS7825I4,** MCS7835I2, MCS7825H3, MCS7825H4, MCS7835H3, MCS7845H3, MCS7845H2, MCS7845I3, MCS7845I2	8.0(2)	Not Applicable	Processing/Signaling
UCS C Series Server **UCS C210-M1 (with VMware)**	8.0(2)		Virtualization
UCS Server **UCS5108 with B200-M1** and B250-M1 (with VMware)	8.0(2)	**6120XP**	Fabric Interconnect
		Cisco MDS 9124	Fiber Channel Switch
		EMC AX4 SAN	Data Storage
Cisco 3845, 3825 Integrated Services Router **Cisco 3945,** 3945E, 3925, 3925E Integrated Service Router Generation 2 (Gateway)	IOS 15.1(1)T	**NM HDV2**	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card. 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		VWIC2-1MFT-T1/E1	First Generation Voice/WAN Interface Card. 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 2.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 2.)
		VIC3 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VIC3 2FXS	Voice Interface Card, 2-port, Foreign Exchange Station
		SM-NM-ADAPTER	Available only in 2951/3945 for SM (Service Module) to NM (Network Module) adaption
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 3.)
		EM3 HDA 8FXS/DID	8-Port HD analog and digital extension module for voice and fax (See note 3.)
		EMV HD 8FXS/DID	8-Port HD analog and digital extension module for voice and fax (See note 3.)
		VIC 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station, DID
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		PVDMII	See note 4.

Table 2-4. SUT Tested System Configurations (continued)

Cisco Unified Communications Manager Version 8.0(2), with IOS Software Release 15.1(1)T			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Function**
Cisco 2951, 2851, 2921, 2911, 2821, and 2811 Integrated Services Router (Gateway)	IOS 15.1(1)T	**NM HD V2**	2-slot IP communications enhanced voice/fax network module
		SM-NM-ADAPTER	Available only in 2951/3945 for SM (Service Module) to NM (Network Module) adaption
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		VWIC2 1MFT T1/E1	First Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		EM3 HD 8FXS/DID	HD analog and digital extension module for voice and fax
		EVM HD 8FXS/DID	HD analog and digital extension module for voice and fax
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 3.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 2.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 3.)
		VIC 4FXS/DID	Voice interface card, 4-port, RJ-11, foreign exchange station, DID
		VIC3 4FXS/DID	Voice interface card, 4-port, RJ-11, foreign exchange station, DID
		VIC 2FXS	Voice Interface card, 2-port, RJ-11, Foreign exchange station
		VIC3 2FXS	Voice Interface card, 2-port, RJ-11, Foreign exchange station
		PVDMII	(See Note 4.)

Table 2-4. SUT Tested System Configurations (continued)

Cisco Unified Communications Manager Version 8.0(2), with IOS Software Release 15.1(1)T			
Component (See note 1.)	**Component (See note 1.)**	**Component (See note 1.)**	**Component (See note 1.)**
VG224	IOS 15.1(1)T		Voice Gateway(VG) for support of up to 24 non secure voice and facsimile analog endpoints only.
3845 CUBE	IOS 15.1(1)T-BORON_116	Not Applicable	AS-SIP Gateway
CP-7940G and CP-7960G (See note 5.)	P00308010200	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7970G and CP-7971G	SCCP70.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7931G	SCCP31.9-0-2SR1S	Not Applicable	IP Phone (with push to talk handset or with standard handset)
CP-7911G and 7906G	SCCP11.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7941G, CP-7941G-GE, CP-7961G, and CP-7961G-GE	SCCP41.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7942G and CP-7962G	SCCP42.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7945G and CP-7965G	SCCP45.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7975G	SCCP75.9-0-2SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7925G	CP7925G-1.3.3.LOADS	Not Applicable	Wireless IP Phone
7914	Load: S00105000400	Not Applicable	Expansion module
7915	B015-1-0-4	Not Applicable	Expansion module
7916	B015-1-0-4	Not Applicable	Expansion module
General Dynamics C4 Systems Sectéra® vIPerTM (See note 7.)	Release 1.0, Software ver.6.04	Not Applicable	IP Phone (with standard handset)
CIS Secure DTD-7961-T-SG-SC-SC-X-X (See note 8.)	SCCP41.9-0-2SR1S	Not Applicable	7961G TEMPEST version with 100 Mbps SC Fiber LAN and PC interfaces, TSG Positive Disconnect, no speakerphone, shared access
CIS Secure DTD-7975-X-XSC-RJ-ME-SE (See note 8.)	SCCP75.9-0-2SR1S	Not Applicable	7975G Standard with 1000 Mbps SC Fiber LAN and RJ45 PC interfaces, shared access
CRYPTEK CT915-V-P1-003 (See note 8.)	SCCP41.9-0-2SR1S	Not Applicable	7961G IP phone, Fiber TEMPEST version with 100MB Fiber LAN and no shared access
Walker WS-2620	Not Applicable	Not Applicable	Push to Talk Handset for Cisco 7900 Series phones
Cisco IP Communicator	7.0.3	Not Applicable	Cisco Softphone Application
The following phones were tested with the SUT, but are not certified for use with the SUT due to critical failures.			
Telecore 2151	2AE-00056-0102	Not Applicable	IP Phone (with push-to-talk handset or with standard handset), 100 Mbps shared access[9]
L-3 Communications IP STE	1.2.4	Not Applicable	IP STE[11]

Table 2-4. SUT Tested System Configurations (continued)

NOTES:
1. Components bolded and underlined were tested by JITC. The other components in the family series were not tested; however, they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.
2. These components are certified in the DSN with T1 ISDN PRI interface. These components are certified in the PSTN with the T1 ISDN PRI and E1 ISDN PRI interfaces.
3. The EM HDA 8FXS and EM3 HDA 8FXS/DID expansion modules require the EVM HD module. Up to two EM HDA 8FXS or EM3 HDA 8FXS/DID expansion modules are supported for each EVM HD.
4. The 2800, 2900, 3800 and 3900 series of Integrated Service Routers (Gateways) are certified with the Packet Voice Digital Signal Processor Module II (PVDMII). The SUT offers a Packet Voice Digital Signal Processor Module 3 (PVDM3) however due to excessive One-Way latency they are not certified.
5. The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in this table and a dual stack call control agent in accordance with the interim UCR change 1 IPv6 rules of engagement, Reference (c).
6. The appropriate certified versions for all gateways-is IOS 15.1(1)T.
7. CIS Secure Computing and Cryptek add security modifications which are physical in nature and do not affect software. The CIS Secure Computing and Cryptek security modifications can be applied to any certified Cisco IP phone.
8 The Telecore 2151 was tested with the SUT; however, it is not certified with the SUT due excessive loss of voice media. The Telecore 2151 failed to meet the Packet Voice Impairment Test (PVIT) requirements when tested with other phones.
9. Calls could not be placed from the L-3 IP STE when it was tested with the SUT. Although the L-3 IP STE certified with previous CUCM versions, it is not certified with this version of the CUCM.

LEGEND:

APL	Approved Product List
AS-SIP	Assured Services Session Initiation Protocol
BIOS	Basic Input Output System
CP	Cisco Phone
DID	Direct Inward Dialing
DISA	Defense Information Systems Agency
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
EM	Expansion Module
EVM	Extension Voice Module
Fax	facsimile
FXS	Foreign Exchange Station
G	10/100BaseT Ethernet (A Cisco part designator on their IP phone.)
GE	Gigabit Ethernet (A Cisco part designator on their IP phone.)
HD	High Density
HDA	High Density Analog
IOS	Internetwork Operating System
IP	Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
JITC	Joint Interoperability Test Command
LAN	Local Area Network
Mbps	Megabits per second
MCS	Media Convergence Server
MFT	Multiflex Trunk
MOS	Mean Opinion Score
NM	Network Module
PC	Personal Computer
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RJ	Registered Jack
SC	fiber connector (square push-in)
SCCP	Skinny Call Control Protocol
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TDM	Time Division Multiplexing
UC	Unified Capabilities
UCR	Unified Capabilities Requirements
UCS	Unified Computing System
V	Voice
VE	Voice/Fax Enhanced
VIC	Voice Interface Card
VWIC	Voice WAN Interface Card
WAN	Wide Area Network

10. TESTING LIMITATIONS. The JITC test team noted the following testing limitations including the impact they may have on interpretation of the results and conclusions. Any untested requirements are also included in the testing limitations.

a. Call Loading. The JITC could not create a large volume of line calls because the line signaling protocol used by the SUT is proprietary. Also, the JITC could not generate a large volume of AS-SIP trunk calls because of limited lines which were provided during testing and lack of AS-SIP test equipment. These limitations pose a low risk to interoperability and should not impact overall results and conclusions. The use of operational data as the LSC is fielded will validate the SUTs ability to support its proposed number of subscribers (up to 30,000).

b. Assured Services Session Initiation Protocol End Instruments. The JITC did not test the SUT with generic AS-SIP End Instruments (AEIs) because none were available at the time of test. A vendor has yet to submit a product as an AEI for certification. This requirement was an addition to the UCR 2008 Change 1 and therefore the SUT has 18-months (July 2011) to comply with the requirement.

c. Proprietary End Instruments. The JITC did not test PEIs for video requirements. Since the Defense Switched Network (DSN) has not deployed videophones under legacy certifications, this poses a low operational impact. The JITC will verify video capabilities of the SUT prior to amending the certification to include the capability.

d. Internet Protocol version 6. The IPv6 requirements were tested in the Private Branch Exchange (PBX) configuration. These results were applied to the LSC configuration where applicable. The JITC did not test IPv6 Inter-enclave (i.e., between LSCs via an Edge Boundary Controller (EBC). The JITC did not test this feature because the EBC did not fully support IPv6 during the time of testing.

e. Network Management. The JITC did not test the SUT's ability to meet UCR NM requirements. The vendor did submit an NM LoC that was reviewed by JITC. The JITC's evaluation of the SUT's NM capabilities is provided in paragraph 11.

f. Attendant Consoles. The JITC did not test the SUT's Attendant features. The vendor did not provide an Attendant Console. This requirement was an addition to the UCR 2008 Change 1 and therefore the SUT has 18-months (July 2011) to comply with the requirement.

g. Master/Slave. The JITC did not test the SUT to determine its ability to meet master/slave requirements. Initial fielding of an LSC will not be used in this configuration. The operational impact was adjudicated to be low.

h. Secure Data and Secure Voice Calls. The standard for modem over IP is based on ITU-T V.150.1 and vendors have 18-months (July 2011) to comply. Secure calls were not tested inter-enclave (between LSCs via DISN). The SUT supports V.150.1 with all its media gateways except the VG224 which is certified only for non secure voice and facsimile. In accordance with UCR 2008 change 1 paragraph 5.3.2.6.1.6 every analog Integrated Access Device (IAD) line card on the LSC or MFSS is not required to support secure voice, secure data, or non secure modem or ITU–T Recommendation V.150.1 Modem Relay.

11. INTEROPERABILITY EVALUATION RESULTS. The SUT meets the critical interoperability requirements for an LSC in accordance with the UCR and is certified for joint use with other UC Products listed on the APL. Additional discussion regarding specific testing results is located in subsequent paragraphs.

11.1 Interfaces. The SUT met line interface requirements for 10/100/1000 Base-X interfaces. These IP line interfaces were met through use of PEIs (voice only). The SUT supports 2-wire analog phones via a gateway. The SUT met the external interface requirements for 10/100/1000Base-X (AS-SIP) , T1Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) for both ANSI T1.619a MLPP and National ISDN-2 (NI-2) commercial, and E1 ISDN PRI for both ITU-T Q.931 and ITU-T Q955.3. The JITC did not test the other conditional interfaces. The interface status of the SUT is provided in Table 2-5.

Table 2-5. SUT Interface Requirements Status

Interface	Critical	UCR Reference	Threshold CR/FR Requirements (See note 1.)	Status	Remarks (See note 2.)
Line Interfaces					
10Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to PEIs and softphones.
100Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to PEIs and softphones.
1000Base-X	No	5.3.2.6.3	2, 4, 10,13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3ab. Applies to PEIs and softphones.
2-wire analog	Yes	5.3.2.6.1.6	2, 4, 10, and 13	Certified	Met threshold CRs/FRs for 2-wire instruments. Applies to 2-wire secure and non-secure analog instruments. Requirement met through use of a Gateway that supports IEEE 802.3i, 802.3u, and 802.3ab (See note 3.).
BRI	No	5.3.2.6.1.8	2, 4, 10, and 13	Not Tested	This interface is offered by the SUT but was not tested because it does not support Assured Services.
External Interfaces					
10Base-X	No (See note 4.)	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to AS-SIP trunk.
100Base-X	No (See note 4.)	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to AS-SIP trunk.
1000Base-X	No (See note 4.)	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3z and 802.3ab. Applies to AS-SIP trunk.
ISDN T1 PRI ANSI T1.619a	Yes	5.3.2.4.3	2, 3, 7, 8, 10, and 13	Certified	Met threshold CRs/FRs . Provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2	Yes	5.3.2.4.3	2, 3, 7, 8, 10, and 13	Certified	Met threshold CRs/FRs. Provides PSTN Connectivity
T1 CCS7 ANSI T1.619a	No	5.3.2.12.9	2 , 3, 7, 8, 10, and 13	Not Tested	This interface is not offered by the SUT.
T1 CAS	No	5.3.2.12.11	2, 3, 7, 8, 10, and 13	Not Tested	This interface is offered by the SUT but was not certified because of known discrepancies (See note 5.).
E1 PRI ITU-T Q.955.3	No (See note 6.)	5.3.2.12.10	2, 3, 7, 8, 10, and 13	Certified	Tested under PBX1 configuration. Results applicable to LSC.
E1 PRI ITU-T Q.931	No (See note 6.)	5.3.2.12.10	2, 3, 7, 8, 10, and 13	Certified	Conditionally required for European PSTN connectivity.

Table 2-5. SUT Interface Requirements Status (continued)

Interface	Critical	UCR Reference	Threshold CR/FR Requirements (See note 1.)	Status	Remarks (See note 2.)
NM					
10Base-X	No (See note 4.)	5.3.2.4.4 5.3.2.7.2.8	16 and 17	Certified	Met threshold CRs/FRs. Verified via LoC.
100Base-X	No (See note 4.)	5.3.2.4.4 5.3.2.7.2.8	16 and 17	Certified	Met threshold CRs/FRs. Verified via LoC.

NOTES:
1. CR/FR requirements are contained in Table 2. CR/FR numbers represent a roll-up of UCR requirements. Enclosure 3 provides a list of more detailed requirements LSC products.
2. Paragraph 11 of Enclosure 2 provides detailed information pertaining to open TDRs and associated operational impacts.
3. Voice calls from the SUT gateway analog interfaces via the UC DISN WAN require a loopback configuration of ANSI T1.619a ISDN PRI interfaces within each gateway (refer to Cisco CUCM deployment guide).   This configuration requires translations in the gateways to route all out going analog calls placed towards the UC DISN WAN via the looped T1s. Additionally, incoming calls from the UC DISN WAN to analog end instruments on each gateway must be routed via the looped T1s. Without this configuration, analog end instruments cannot place calls via the UC DISN WAN. This configuration requires two looped ISDN PRI ANSI T1.619a T1s within each 3845 and 3945 gateways and will support a maximum of 69 analog interfaces per gateway.   This allows for up to two ISDN PRI T1 interfaces or one ISDN PRI E1 interface for timing/network access.   In addition, each 2851 and 2951 gateway requires one looped ANSI T1.619a ISDN PRI within each 2851 and 2951 gateway and will support a maximum of 23 analog interfaces per gateway. Both gateways also require a T1 or E1 interface for synchronization via recovered timing.
4. Must provide a minimum of one of the listed interfaces.
5. The SUT CAS interface had interoperability test discrepancies adjudicated to be critical for certification of this interface.
6. The interface is conditionally required for deployment in Europe.

LEGEND:

ANSI	American National Standards Institute
AS-SIP	Assured Services Session Initiation
BRI	Basic Rate Interface
CAS	Channel Associated Signaling
CCS7	Common Channel Signaling 7
CR	Capability Requirement
CUCM	Cisco Unified Communications Manager
DISN	Defense Information System Network
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
FR	Functional Requirement
IEEE	Institute of Electrical and Electronics Engineers, Inc.
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunications Union Telecommunication Standardization Sector
LoC	Letter of Compliance
LSC	Local Session Controller
Mbps	Megabits per second
NI-2	National ISDN Standard 2
NM	Network Management
PEI	Proprietary End Instrument
PRI	Primary Rate Interface
PSTN	Public Switch Telephone Network
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TDR	Test Discrepancy Report
UCR	Unified Capabilities Requirements
WAN	Wide Area Network

11.2 Capability Requirements (CR) and Functional Requirements (FR). The SUT CR and FR status is depicted in Table 2-6. Detailed CR/FR requirements are provided in Enclosure 3, Table 3-1. A summary of the SUT's ability to meet UCR requirements are provided in the sub-paragraphs below. All requirements and associated references were derived from UCR 2008 Change 1. Discrepancies discussed below were adjudicated to be minor based on vendor submission and compliance to a Plan of Actions and Milestones.

Table 2-6. SUT Capability Requirements and Functional Requirements Status

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Status	Remarks
1	**Assured Services Product Features and Capabilities**				
	DSCP Packet Marking	Required	5.3.2.2.1.4	Met	
	Voice Features and Capabilities	Required	5.3.2.2.2.1	Partially Met	See note 2.
	Public Safety Features	Required	5.3.2.2.2.2	Met	
	ASAC – Open Loop	Required	5.3.2.2.2.3	Met	
	Signaling Protocols	Required	5.3.2.2.2.3	Met	
	Signaling Performance	Required	5.3.2.2.2.4	Met	
2	**Registration, Authentication, and Failover**				
	Registration	Required	5.3.2.3.1	Met	
	Failover	Required	5.3.2.3.2	Met	
3	**Product Physical, Quality, and Environmental Factors**				
	Availability	Required	5.3.2.5.2.1	Partially Met	See note 3
	Maximum Downtimes	Required	5.3.2.5.2.2	Met	
	Loss of Packets	Required (See note 4.)	5.3.2.5.4	Met	
4	**Voice End Instruments**				
	Tones and Announcements	Required	5.3.2.6.1.1	Partially Met	See notes 2 and 5.
	Audio Codecs	Required	5.3.2.6.1.2	Partially Met	See note 5.
	VoIP PEI or AEI Audio Performance Requirements	Required	5.3.2.6.1.3	Partially Met	See note 5.
	VoIP Sampling Standard	Required	5.3.2.6.1.4	Partially Met	See note 5.
	Authentication to LSC	Required	5.3.2.6.1.5	Partially Met	See note 5.
	Analog Telephone Support	Required (See note 6.)	5.3.2.6.1.6	Partially Met	See note 7.
	Softphones	Conditional	5.3.2.6.1.7	Met	See note 8.
	ISDN BRI	Conditional	5.3.2.6.1.8	Not Tested	
5	**Video End Instruments**				
	Video End Instrument	Required	5.3.2.6.2	Not Tested	See note 8.
	Display Messages, Tones, and Announcements	Required	5.3.2.6.2.1	Not Tested	See note 8.
	Video Codecs (Including Associated Audio Codecs)	Required	5.3.2.6.2.2	Not Tested	See note 8.
6	**LSC Requirements**				
	PBAS/ASAC Requirements	Required	5.3.2.7.2.1	Met	
	Calling Number Delivery Requirements	Required	5.3.2.7.2.2	Met	
	LSC Signaling Requirements	Required	5.3.2.7.2.3	Met	
	Service Requirements under Total Loss of WAN Transport	Required	5.3.2.7.2.4	Met	
	Local Location Server and Directory	Required	5.3.2.7.2.5	Met	
	LSC Transport Interface Functions	Required	5.3.2.7.2.7	Met	
	LSC to PEI, AEI, and Operator Console Status Verification	Required	5.3.2.7.2.10	Partially Met.	See note 9.
	Line-Side Custom Features Interference	Conditional	5.3.2.7.2.11	Met	
	Loop Avoidance	Required (See note 4.)	5.3.2.7.3	Met	

Table 2-6. SUT Capability Requirements and Functional Requirements Status (continued)

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Status	Remarks
7	**Call Connection Agent Requirements**				
	CCA IWF Component	Required (See note 10.)	5.3.2.9.2.1	Met	See note 11.
	CCA MGC Component	Required (See note 10.)	5.3.2.9.2.2	Met	
	SG Component	Conditional	5.3.2.9.2.3	Not Tested	See note 11.
	CCA-IWF Support for AS-SIP	Required	5.3.2.9.5.1	Met	
	CCA-IWF Support for SS7	Conditional	5.3.2.9.5.2	Not Tested	See note 11.
	CCA-IWF Support for PRI via MG	Required	5.3.2.9.5.3	Met	
	CCA-IWF Support for CAS Trunks via MG	Conditional	5.3.2.9.5.4	Not Tested	See note 11.
	CCA-IWF Support for PEI and AEI Signaling Protocols	Required	5.3.2.9.5.5	Partially Met	See note 12.
	CCA-IWF Support for VoIP and TDM Protocol Interworking	Required (See note 10.)	5.3.2.9.5.6	Met	See note 11.
	CCA Preservation of Call Ringing State during Failure Conditions	Required (See note 4.)	5.3.2.9.6	Met	
	CCA Interactions with Transport Interface Functions	Required	5.3.2.10.3	Met	
	CCA Interactions with the EBC	Required	5.3.2.10.4	Met	
	CCA Support for Admission Control	Required	5.3.2.10.5	Met	
	CCA Support for UFS	Required	5.3.2.10.6	Met	
	CCA Support for IA	Required	5.3.2.10.7	Met	
	CCA Interaction with EIs	Required	5.3.2.10.10	Partially Met	See note 5.
	CCA Support for AS Voice and Video	Required	5.3.2.10.11	Partially Met	See note 8.
	CCA Interactions with Service control Functions	Required	5.3.2.10.12	Met	
	CCA Interworking between AS-SIP and SS7	Conditional	5.3.2.11	Not Tested	See note 11.
8	**MG Requirements**				
	Role of MG In LSC	Required	5.3.2.12.3.1	Met	
	MG Support for ASAC	Required	5.3.2.12.4.1	Met	
	MG and IA Functions	Required	5.3.2.12.4.2	Met	
	MG Interaction with Service Control Function	Required	5.3.2.12.4.3	Met	
	MG Interactions with IP Transport Interface Functions	Required	5.3.2.12.4.4	Met	
	MG-EBC interactions	Required	5.3.2.12.4.5	Met	
	MG IP-Based PSTN Interface Requirements	Conditional	5.3.2.12.4.7	Not Tested	
	MG Interaction with EIs	Required	5.3.2.12.4.8	Met	See note 5.
	MG support for User Features and Services	Required	5.3.2.12.4.9	Met	
	MG Interface to TDM	Required	5.3.2.12.5	Met	See notes 10 and 11.
	MG Interface to TDM Allied and Coalition	Conditional	5.3.2.12.6	Not Tested	
	MG Interface to TDM PSTN in US	Required	5.3.2.12.7	Met	See note 10.
	MG Interfaces to TDM PSTN OCONUS	Required	5.3.2.12.8	Met	See note 10.
	MG Support for CCS7	Conditional	5.3.2.12.9	Not Tested	See note 11.
	MG Support for ISDN PRI Trunks	Required	5.3.2.12.10	Met	

Table 2-6. SUT Capability Requirements and Functional Requirements Status (continued)

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Status	Remarks
8	**MG Requirements (continued)**				
	MG Support for CAS Trunks	Required	5.3.2.12.11	Not Tested	See note 11.
	MG requirements for VoIP Internal Interfaces	Required	5.3.2.12.12	Met	
	MG Echo Cancellation	Required	5.3.2.12.13	Met	
	MG Clock Timing	Required	5.3.2.12.14	Met	
	MGC-MG CCA Functions	Required	5.3.2.12.15	Met	
	MG V.150.1	Required	5.3.2.12.16	Not tested	See note 7.
	MG Preservation of Call Ringing during Failure	Required (See note 4.)	5.3.2.12.17	Met	
9	**SG Requirements**				
	SG and CCS7 Network Interactions	Conditional	5.3.2.13.5.1	Not Tested	See note 11.
	SG Interactions with CCA	Conditional	5.3.2.13.5.2	Not Tested	See note 11.
	SG Interworking Functions	Conditional	5.3.2.13.5.3	Not Tested	See note 11.
10	**WWNDP Requirements**				
	WWNDP	Required	5.3.2.16	Met	
	DSN WWNDP	Required	5.3.2.16.1	Met	
11	**Commercial Cost Avoidance**				
	Commercial Cost Avoidance	Required (See note 3.)	5.3.2.23	Not Tested	
12	**AS-SIP Based for External Devices (Voicemail, Unified Messaging, and Automated Receiving Devices)**				
	AS-SIP Requirements for External Interfaces	Conditional	5.3.2.24	Not Tested	
13	**Precedence Call Diversion**				
	Precedence call Diversion	Required	5.3.2.25	Met	
14	**Attendant Station Features**				
	Precedence and Preemption	Required (See note 3.)	5.3.2.26.1	Not Tested	See note 13.
	Call Display	Required (See note 3.)	5.3.2.26.2	Not Tested	See note 13.
	Class of Service Override	Required (See note 3.)	5.3.2.26.3	Not Tested	See note 13.
	Busy Override and Busy Verification	Required (See note 3.)	5.3.2.26.4	Not Tested	See note 13.
	Night service	Required (See note 3.)	5.3.2.26.5	Not Tested	See note 13.
	Automatic Recall of Attendant	Required (See note 3.)	5.3.2.26.6	Not Tested	See note 13
	Calls in Queue to the Attendant	Required (See note 3.)	5.3.2.26.7	Not Tested	See note 13.
15	**AS-SIP Requirements**				
	SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs	Required (See note 3.)	5.3.4.7	Not Tested	
	SIP Session Keep-Alive Timer	Required	5.3.4.8	Met	
	Session Description Protocol	Required	5.3.4.9	Met	
	Precedence and Preemption	Required	5.3.4.10	Met	
	Video Telephony – General Rules	Required	5.3.4.12	Not Tested	See note 8.
	Calling Services	Required	5.3.4.13	Met	
	SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances	Required	5.3.4.14	Met	
	Relevant Timers for the Terminating Gateway and the Originating Gateway	Required	5.3.4.15	Met	

Table 2-6. SUT Capability Requirements and Functional Requirements Status (continued)

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference	Status	Remarks
15	**AS-SIP Requirements (continued)**				
	SIP Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.16	Met	
	Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.17	Met	
	Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances	Required	5.3.4.18	Met	
	Supplementary Services	Required	5.3.4.19	Met	
16	**IPv6 Requirements**				
	Product Requirements	Required	5.3.5.4	Partially met	See note 14.
17	**NM**				
	LSC Management Function	Required	5.3.2.7.2.6	Met	
	VVoIP NMS Interface Requirements	Required	5.3.2.4.4	Met	
	General Management requirements	Required	5.3.2.17.2	Partially Met	See note 15.
	Requirement for FCAPS Management	Required	5.3.2.17.3	Partially Met	See note 15.
	NM requirements of Appliance Functions	Required	5.3.2.18	Partially Met	See note 15.
	Accounting Management	Required	5.3.2.19	Met	See note 16.

NOTES:
1. Annotation of ‘required’ refers to high level requirement category. Applicability of each sub-requirement is provided in enclosure 3.
2. The SUT had outstanding open TDRs at the completion of testing adjudicated by DISA to have a minor operational impact. The vendor has submitted a PoAM to address the open TDRs. Paragraph 11 of Enclosure 2 provides additional details.
3. When the SUT fails from the primary processor to backup processor all active drop after approx 6-8 minutes. DISA adjudicated this TDR as minor with the vendor’s submitted PoAM to fix by June 2011.
4. This requirement represents a new UCR requirement where the vendor has 18-months (July 2011) to comply.
5. SUT met the requirement for PEIs; SUT was not tested with generic AEI because no AEI was provided. AEIs are a new UCR 2008 Change 1 requirement; the vendor has 18-months (July 2011) to comply.
6. UCR 2008 Change 1 added 18-month rule for G.711 and V.150.1 IAD support.
7. Vendor submitted LoC stating compliance to V.150 however this feature could not be tested because it is not supported by other vendors. This is a new UCR 2008 change 1 requirement; therefore the vendor has until July 2011 to comply with this requirement.
8. SUT did not demonstrate video requirements (conditional for softphone). Vendor did not provide a PEI video capability. This was adjudicated by DISA to have a low operational impact because of the limited deployment of PEIs with video.
9. SUT partially met PEI requirements (no video). The AEI and Operator Console requirements were not tested; the 18-month rule for complying (July 2011) applies.
10. The SUT must meet T1 PRI (T1.619a and NI-2) IWF. The T1 CAS and T1 CCS7 are conditional, and E1 ISDN PRI is conditional for deployment in Europe.
11. The SUT met T1/E1 PRI IWF requirements. The T1 CAS is supported but not certified and T1 CCS7 is not supported by the SUT.
12. The SUT met PEI CCA-IWF requirements. The AEI CCA-IWF requirements were not tested. The 18-month rule applies to AEIs.
13. The Attendant Console requirements are new UCR requirements; 18-month rule applies.
14. The SUT submitted an IPv6 LoC with noted discrepancies. Open TDRs were adjudicated by DISA to have a minor operational impact with vendor submitted PoAM.
15. The SUT submitted an NM LoC with noted discrepancies. Open TDRs were adjudicated by DISA to have a minor operational impact with vendor submitted PoAM.
16. The SUT does not comply with the objective requirement for Record Format.

Table 2-6. SUT Capability Requirements and Functional Requirements Status (continued)

LEGEND:

AEI	AS-SIP End Instrument	Mbps	Megabits per second
AS	Assured Services	MG	Media Gateway
ASAC	Assured Services Admission Control	MGC	Media Gateway Controller
AS-SIP	Assured Services Session Initiation Protocol	NI-2	National ISDN Standard 2
BRI	Basic Rate Interface	NM	Network Management
CCA	Call Control Agent	NMS	Network Management System
CCS7	Common Channel Signaling 7	OCOUNUS	Outside the Continental United States
CR	Capabilities Requirement	PBAS	Precedence-Based Assured Service
DISA	Defense Information System Agency	PEI	Proprietary End Instrument
DSCP	Differentiated Services Code Point	PoAM	Plan of Action and Milestones
DSN	Defense Switched Network	PRI	Primary Rate Interface
EBC	Edge Boundary Controller	PSTN	Public Switch Telephone Network
FCAPS	Fault, Configuration, Accounting, Performance and Security	SG	Signaling Gateway
		SIP	Session Initiation Protocol
FR	Functional Requirement	SS7	Signaling System Number 7
IA	Information Assurance	SUT	System Under Test
IAD	Integrated Access Device	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ID	Identification	TDM	Time Division Multiplexing
IP	Internet Protocol	UCR	Unified Capabilities Requirements
IPv6	Internet Protocol version 6	UFC	User Features and Services
ISDN	Integrated Services Digital Network	US	United States
IWF	Interworking Function	VoIP	Voice over Internet Protocol
JITC	Joint Interoperability Test Command	VVoIP	Voice and Video over Internet Protocol
LoC	Letter of Compliance	WAN	Wide Area Network
LSC	Local Session Controller	WWNDP	World Wide Numbering and Dialing Plan

a. Assured Services Product Features and Capabilities.

(1) DSCP Packet Marking. As part of the session setup process, the LSC controls what DSCP to use in the subsequent session media stream packets. The exact DSCP method used shall comply with Section 5.3.3.3.2. The SUT met all DSCP Packet Marking requirements for IPv4 and IPv6 (intra-enclave only).

(2) Voice Features and Capabilities. The LSC must provide all of the features listed in Table 5.3.2.2-1 of the UCR. The SUT met all Voice Features and Capabilities requirements, with the following exception: SUT does not provide “Ping-Ring” on a call forward variable enabled phone. This was adjudicated as having a minor operational impact. Based on this finding, DISA is revising the UCR to make “Ping-Ring” on a phone assigned call forward variable conditional.

(3) Public Safety Features. The LSC must provide basic emergency service (911), tracing of terminating calls, outgoing call tracing, and tracing of a call in progress. The SUT met all Public Safety Features requirements with the following exception: SUT does not support Trace of a call in progress or Tandem call tracing. This was adjudicated as having a minor operational impact.

(4) ASAC – Open Loop. The LSC must meet the ASAC requirements for the LSC and the MFSS. In the execution of ASAC, certain procedures need to be followed, such as (a) actions to be taken if a precedence session request cannot be completed

because existing sessions are at equal or higher precedence, or (b) tones to be generated when a session is preempted. The SUT met all ASAC requirements.

(5) Signaling Protocols. The LSC must use appropriate signaling for specific trunk types. The control/management protocol between the PEI and the LSC is, in general, proprietary. The control/management protocol between the AEI and the LSC is AS-SIP as specified in Section 5.3.4, AS-SIP Requirements, of this document. The signaling protocol used on UC IP trunks is AS-SIP as specified in Section 5.3.4, AS-SIP Requirements. The MG in the LSC uses ANSI T1.619a PRI or E1 (Q.955.3) signaling on DSN PRI trunks. The SUT met all Signaling Protocol requirements except for AEI. The conditional requirement for CAS was not tested.

(6) Signaling Performance. The LSC has conditional requirements for call setup and tear-down times. SUT met all signaling performance requirements

b. Registration, Authentication, and Failover.

(1) Registration. Registration and authentication between the LSC and EIs shall follow the requirements set forth in UCR 2008, Section 5.4, Information Assurance Requirements. This feature is tested by IA and is covered in the IA report (see paragraph 11.3).

(2) Failover. The LSCs shall be registered to a primary and backup MFSS. In case of failure of the primary MFSS, the LSC will default to the backup MFSS. The SUT met all failover requirements with the following exception. All active calls failed 6-8 minutes after processor failover because the SUT fails to reply to AS-SIP keep alive messages from the destination. The DISA adjudicated this discrepancy as having a minor operational impact due to limited fielding of LSCs within the operational network and is predicated on the vendor's POAM to fix this discrepancy by June 2011.

c. Product Physical, Quality, and Environmental Factors.

(1) Availability. The Assured Services subsystem shall have a hardware/software availability of 0.99999 (non-availability of no more than 5 minutes per year).   This requirement was met via vendor LOC. The SUT met all component failover requirements with the following exception. All active calls 6-8 minutes after failover are disconnected because the SUT fails to reply to AS-SIP keep alive messages from the destination. The DISA adjudicated this discrepancy as having a minor operational impact due to limited fielding of LSCs within the operational network and is predicated on the vendor's POAM to fix this discrepancy by June 2011.

(2) Maximum Downtimes.   The performance parameters associated with the ASLAN, MFSS, and LSC, when combined, shall meet the following maximum downtime requirements:

• IP (10/100 Ethernet) network links – 35 minutes/year

• IP subscriber – 12 minutes/year

This requirement was met via vendor LOC.

(3) Loss of Packets. For VoIP devices, the voice quality shall have a MOS of 4.0 (R-Factor equals 80) or better, as measured in accordance with the E-Model. Additionally, these devices shall not lose two or more consecutive packets in a minute and shall not lose more than seven voice packets (excluding signaling packets) in a 5-minute period. The SUT met all Packet Loss requirements for PEIs.

d. Voice End Instrument. The SUT met PEI requirements except for the noted discrepancies in the following sub-paragraphs. The SUT was not tested with an AEI because no AEIs have been submitted through the UC process. The AEI requirement was a new requirement for which the 18-month rule (July 2011) applies.

(1) Tones and announcements. Tones and announcements, as required in UCR 2008, Sections 5.2.4.5.2, DSN Information Signals, and Section 5.2.2.1.3, Announcements, shall be supported, except for the loss of C2 announcement. The SUT met all requirements for tones and announcements for PEIs.

(2) Audio codecs. The LSC shall support the origination and termination of a voice session using the following codecs: G.711 (a-law and µ-law), G.723.1, G.729 or G.729A, and G.722.1. The SUT met all audio codecs requirements with the following exception: The SUT does not support G.723.1 codec on PEIs. This was adjudicated as having minor operational impact. Based on this finding, DISA is revising the UCR to make the G.723.1 codec a conditional requirement for PEIs and AEIs.

(3) VoIP PEI or AEI Audio Performance Requirements. Voice over IP PEIs or AEIs (i.e., handset, headset, and hands-free types) shall comply with TIA-810-B, November 3, 2006. The SUT met all audio performance requirements for PEIs.

(4) VoIP Sampling Standard. For Fixed-to-Fixed calls, the product shall use 20 ms as the default voice sample length, and as the basis for the voice payload packet size. The SUT met the VoIP sampling standard requirements for PEIs.

(5) Authentication to LSC. The PEI and AEI shall be capable of authenticating itself to its associated LSC and vice versa. The SUT met all PEI to LSC authentication requirements.

(6) Analog Telephone Support. Analog instruments, including secure analog EIs, analog facsimile EIs, and analog modem EIs, shall be supported by the LSC either by a Terminal Adapter or an Integrated Access Device (IAD) connected to an Ethernet port. The SUT met all analog telephone support requirements via an IAD integrated within their media gateway and a standalone VG224 analog gateway.

(7) Softphones. The softphone shall be conceptually identical to a traditional

IP “hard” telephone and is required to provide voice features and functionality provided by a traditional IP hard telephone. The SUT meet all softphones requirements.

(8) ISDN BRI. The ISDN BRI EIs, including secure ISDN BRI EIs, may be supported by the LSC. This is a conditional requirement; no BRI EIs were provided on the SUT at the time of test.

(9) The Telecore 2151 was tested with the SUT; however, it is not certified with the SUT due excessive loss of voice media. The Telecore 2151 failed to meet the Packet Voice Impairment Test (PVIT) requirements when tested with other phones.

(10) Calls could not be placed from the L-3 IP STE when it was tested with the SUT. Although the L-3 IP STE certified with previous CUCM versions, it is not certified with this version of the CUCM.

e. Video End Instruments. The SUT must support both voice and video. PEIs and AEIs can support only voice, only video or both voice and video. The SUT’s PEI support voice only and did not support video. This discrepancy was adjudicated to have a minor operational impact because LSC managed video represents a new requirement not fielded under previous time division multiplexing (TDM) certifications. The SUT was not tested with an AEI. The AEI requirement was a new requirement for which the 18-month rule (July 2011) applies.

(1) Video End Instrument. Video EIs are considered associated with the LSC and must have been designed in conjunction with the LSC design. An IP video instrument shall be designed in accordance with the acquiring activity requirements. This was not tested as the SUT did not provide any video end instruments.

(2) Display Messages, Tones, and Announcements. Tones and announcements, as appropriate for voice and video over IP, and as required, in UCR 2008, Sections 5.2.4.5.2, DSN Information Signals, 5.2.2.1.3, Announcements, shall be supported by the PEI and AEI. This was not tested as the SUT did not provide any video end instruments.

(3) Video Codecs (Including Associated Audio Codecs). The product shall support the origination, maintenance, and termination of a video session using the following codecs: one G.xxx and one H.xxx must be used to create and sustain a video session. This was not tested as the SUT did not provide any video end instruments.

f. LSC Requirements.

(1) PBAS/ASAC Requirements. The LSC shall meet all the requirements for PBAS/ASAC, as appropriate for VoIP and Video over IP services, as specified in UCR 2008, Section 5.2.2, Multilevel Precedence and Preemption. The SUT met all PBAS/ASAC Requirements with the following exception. The SUT does not provide directionalization of its ASAC budget. This discrepancy was adjudicated to have a

minor operational impact.

(2) Calling Number Delivery Requirements. The calling number provided to the called party shall be determined by the dial plan serving the calling instrument in accordance with Telcordia Technologies GR-31-CORE *"CLASS℠ Feature: Calling Number Delivery,"* Issue 1, June 2000. The SUT met all calling number delivery requirements.

(3) LSC Signaling Requirements. The LSC must provide signaling on the line side for local intra-enclave subscriber-to-subscriber calls, and trunk-side signaling for calls between an external enclave and a local subscriber. The SUT met all LSC Signaling Requirements.

(4) Service Requirements under Total Loss of WAN Transport. In the event that a total loss of connectivity to the DISN WAN occurs, the LSC shall provide the following functions:

- Completion of local (intra-enclave) calls
- Routing of calls to the PSTN using a local MG (PRI or CAS as required by the local interface)
- User look-up of local directory information

The SUT met all Service Requirements under Total Loss of WAN Transport.

(5) Local Location Server and Directory. The purpose of the Local Location Server (LLS) is to provide information on call routing and called address translation (where a called address is contained within the called SIP URI in the form of the called number). The SUT met all LLS and directory requirements.

(6) LSC Transport Interface Functions. The LSC Transport Interface functions provide interface and connectivity functions with the ASLAN and its IP packet transport network. The SUT met all transport interface function requirements.

(7) LSC to PEI, AEI, and Operator Console Status Verification. Periodically, the LSC shall verify the status of its registered and authenticated IP EIs. The SUT met all status verification requirements for PEIs. The SUT was not tested with AEIs and Attendant Console. Both are new UCR requirements for which the 18-month rule applies.

(8) Line-Side Custom Features Interference. Vendors may implement unique custom features applicable to the line side of the LSC. Line-side custom features must not interfere with the Assured Services requirements. The SUT offers line-side custom features. However, JITC did not test any of those features; therefore, they are not certified for use.

(9) Loop Avoidance. During the call establishment process, the product shall

be capable of preventing or detecting and stopping hair-pin routing loops over ANSI T1.619a and commercial PRI trunk groups (i.e., T1 PRI and E1 PRI) between a legacy switch (e.g., TDM EO) and an LSC. The SUT met all Loop Avoidance requirements for T1 PRI (ANSI T1.619a and NI-2) and E1 PRI (ITU-T Q.931 and ITU-T Q.955.3).

g. Call Connection Agent Requirements.

(1) CCA IWF Component. The role of the IWF within the CCA is to support all the VoIP and TDM signaling protocols that the LSC supports for EIs, MGs, and EBCs, and to Interwork all these various signaling protocols with one another. The SUT met all CCA IWF requirements for T1 PRI (ANSI T1.619a and NI-2) and E1 PRI (ITU-T Q.931 and ITU-T Q.955.3). Of the other conditional IWFs, T1 CAS was tested but had discrepancies adjudicated by DISA to be critical and T1 CCS7 which is not offered by the SUT.

(2) CCA MGC Component. The MGC within the CCA must control all MGs within the LSC or MFSS, control all trunks within each MG, control all signaling and media streams on each trunk within each MG, accept IP-encapsulated signaling streams from an SG or MG, and to use either ITU-T recommendation H.248 or a supplier-proprietary protocol to accomplish these controls. The SUT met all CCA MGC requirements.

(3) SG Component. The role of the CCA with respect to the SG is to control all SGs within the network appliance, and to control all signaling links (DoD CCS7) within each SG. The SG is conditional for an LSC and was not tested on the SUT.

(4) CCA-IWF Support for AS-SIP. The CCA IWF shall support the AS-SIP protocol consistent with the detailed AS-SIP protocol requirements in Section 5.3.4, AS-SIP Requirements. The SUT met all requirements for CCA-IWF support for AS-SIP for required interfaces (T1 PRI (ANSI T1.619a and NI-2) and E1 PRI (ITU-T Q.931). Of the other conditional IWFs, T1 CAS was tested but had discrepancies adjudicated by DISA to be critical and T1 CCS7 was not tested.

(5) CCA-IWF Support for SS7. CCA-IWF support for SS7 is a conditional requirement for LSCs and is not supported by the SUT.

(6) CCA-IWF Support for PRI, via MG. The CCA IWF shall support the U.S./National ISDN version of the ISDN PRI protocol. The SUT met all requirements for CCA-IWF support for T1 PRI (ANSI T1.619a and NI-2) and E1 PRI (ITU-T Q.931 and ITU-T 955.3).

(7) CCA-IWF Support for CAS Trunks via MG. Support for CAS is a conditional requirement for LSCs. The CAS interface was tested on the SUT but not certified because of discrepancies adjudicated by DISA to be critical.

(8) CCA-IWF Support for PEI and AEI Signaling Protocols. The CCA IWF

shall support supplier-proprietary Voice and Video EIs and their associated proprietary EI signaling protocols. The SUT met all requirements for CCA-IWF Support for PEI Signaling Protocols. No AEIs were tested.

(9) CCA-IWF Support for VoIP and TDM Protocol Interworking. The role of the IWF within the CCA is to support all the VoIP and TDM signaling protocols that the appliance supports for PEIs, AEIs, MGs, and EBCs, and interwork all these various signaling protocols with one another. The SUT met all requirements for CCA-IWF Support for VoIP and TDM Protocol Interworking required interfaces (T1 PRI (ANSI T1.619a and NI-2) and E1 PRI (ITU-T Q.931 and ITU-T Q.955.3). Of the other conditional IWFs, T1 CAS was tested but had discrepancies adjudicated by DISA to be critical and T1 CCS7 was not tested.

(10) CCA Preservation of Call Ringing State during Failure Conditions. The CCA in the LSC, MFSS, and WAN SS shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within the CCA. This requirement was not tested. This is a new UCR requirement for which the 18-month rule applies.

(11) CCA Interactions with Transport Interface Functions. The CCA interacts with Transport Interface functions by using them to communicate with PEIs, AEIs, the EBC, the MGs, and the SG over the ASLAN. The SUT met all requirements for CCA interactions with Transport Interface Functions with the exception of AEIs. No AEIs were tested.

(12) CCA Interactions with the EBC. The CCA interacts with the EBC by directing AS-SIP signaling packets to it (for signaling messages destined for an MFSS) and by accepting AS-SIP signaling packets from it (for signaling messages directed to the LSC from an MFSS). The SUT met all requirements for CCA interactions with the EBC.

(13) CCA Support for Admission Control. The CCA interacts with the ASAC component of the LSC and MFSS to perform specific functions related to ASAC, such as counting internal, outgoing, and incoming calls; managing separate call budgets for VoIP and Video over IP calls; and providing preemption. The SUT met all requirements for CCA support for Admission Control

(14) CCA Support for UFS. The UFS Server is responsible for providing features and services to VoIP and Video PEIs/AEIs on an LSC or MFSS, where the CCA alone cannot provide the feature or service. The SUT met all requirements for CCA Support for UFS for PEIs.

(15) CCA Support for IA. The Information Assurance function within the appliance ensures that end users, PEIs, AEIs, MGs, SGs, and EBCs that use the

appliance are all properly authenticated and authorized by the appliance. The Information Assurance function ensures that Voice and Video signaling streams that traverse the appliance and its ASLAN are encrypted properly SIP/TLS. IA requirements are tested separately; see paragraph 11.3.

(16) CCA Interaction with EIs. The LLS provides information on called address translation in response to call routing queries from the CCA. The CCA sends call routing queries to the LLS for both outgoing calls from appliance PEIs or AEIs (i.e., LSC and MFSS) and incoming calls to appliance PEIs or AEIs (i.e., LSC and MFSS). The SUT met all requirements for CCA interaction with PEIs.

(17) CCA Support for AS Voice and Video. The CCA in the MFSS or LSC needs to interact with VoIP PEIs and AEIs served by that MFSS or LSC. The VoIP interface between the PEI and the MFSS or LSC is left up to the network appliance supplier. The VoIP interface between the AEI and the MFSS or LSC is AS-SIP. The SUT met all requirements for CCA support for AS Voice. The SUT did not provide Video. This was adjudicated to have a minor operational impact.

(18) CCA Interactions with Service Control Functions. The CCA shall support the ability to remove VoIP and Video sessions and session requests from the media server so the CCA can continue with necessary session processing once the media server has completed its functions. The SUT met all requirements for CCA Interactions with Service Control Functions.

(19) CCA Interworking between AS-SIP and CCS7. Interworking is performed at a node with CCA (SIP/CCS7 IWF) functionality that processes/interworks incoming CCS7 messages to outgoing AS-SIP messages, and similarly, incoming AS-SIP messages to outgoing CCS7 messages. This is a conditional requirement for LSCs and was not tested on the SUT.

h. MG Requirements.

(1) Role of MG In LSC. The MG supports interconnection of VoIP, FoIP, and MoIP media streams with the LSC media server, which provides tones and announcements for LSC calls and LSC features. To support inter-enclave MoIP and FoIP, the LSC must meet ITU-T V.150.1 requirements. The V.150.1 requirement is a new UCR requirement for which the 18-month (July 2011) rule applies.

(2) MG Support for ASAC. The MG assists the CCA in performing ASAC (i.e., call preemption based on per-call precedence levels) for outgoing TDM calls at MGs and for incoming TDM calls at MGs. The SUT met all requirements for MG Support for ASAC.

(3) MG and IA Functions. The Information Assurance function within the appliance ensures that end users, PEIs, AEIs, MGs, SGs, and EBCs that use the appliance are all properly authenticated by the appliance. The Information Assurance

function also ensures that VoIP signaling streams and media streams that traverse the appliance and its ASLAN are properly encrypted, using SIP/TLS and SRTP, respectively. IA requirements are tested separately; see IA report.

(4) MG Interaction with Service Control Function. The MG is responsible for routing individual VoIP, FoIP, and MoIP media streams to the media server when instructed to do so by the CCA/MGC. When instructed to do so by the CCA/MGC, the MG is responsible for removing individual VoIP, FoIP, and MoIP media streams from the media server, and for either disconnecting them entirely, or routing them on to other LSC end users (e.g., VoIP or video EIs). The SUT met all requirements for MG Interaction with Service Control Function except for V.150.1. The V.150.1 requirement is a new UCR requirement for which the 18-month (July 2011) rule applies.

(5) MG Interactions with IP Transport Interface Functions. The Transport Interface functions in the LSC provide interface and connectivity functions with the ASLAN and its IP packet transport network. The SUT met all requirements for MG Interactions with IP Transport Interface Functions.

(6) MG-EBC interactions. The MG interacts with the EBC by sending SRTP media streams to it (for call media destined for a PEI, AEI, or MG that is served by another appliance outside the LSC), or by accepting SRTP media streams from it (for call media arriving from a PEI, AEI, or MG that is served by another appliance outside the LSC). The SUT met all requirements for MG-EBC interactions for PEI interactions.

(7) MG IP-Based PSTN Interface Requirements. Voice and Video over IP interfaces from the UC network to the PSTN have not been defined. Interfaces from an LSC or MFSS to the PSTN will be via an MG with TDM interfaces as specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features.

(8) MG Interaction with EIs. The MG in the MFSS or LSC needs to interact with VoIP EIs served by that MFSS or LSC, and with VoIP EIs served by other MFSSs or LSCs. The VoIP signaling interface between the PEI and the MFSS or LSC is left up to the network appliance supplier. The VoIP signaling interface between the AEI and the MFSS or LSC is AS-SIP. The SUT met all MG Interaction with VoIP EIs requirements with the following exception. The PVDM3 Digital Signal Processor in the 2951 and 3945 gateways exceeded the one-way latency requirements measured from the IP handset to the T1 egress. The PVDM2 processors met all other requirements. This was adjudicated has having minor operational impact.

(9) MG Support for User Features and Services. The MG shall support the operation of features for VoIP and Video end users, consistent with the operation of this feature on analog and ISDN lines in DoD TDM switches today. The SUT met all requirements for MG Support for User Features and Services.

(10) MG Interface to TDM network elements in DoD Networks. Each appliance MG shall support TDM trunk groups that can interconnect with the following

devices in DoD networks, in the United States and worldwide: PBXs, SMEOs, EOs, and MFSS. The SUT met all requirements for MG Interface to TDM devices in DoD Networks.

(11) MG Interface to TDM Allied and Coalition. The appliance suppliers should support TDM trunk groups on their MG product that can interconnect with devices in U.S. allied and coalition partner networks worldwide. This requirement is conditional and was not tested on the SUT.

(12) MG Interface to TDM PSTN in US. Each appliance MG shall support TDM trunk groups that can interconnect with NEs in the PSTN in the United States. The SUT met all requirements for MG Interface to TDM PSTN in the US using T1 PRI.

(13) MG Interfaces to TDM PSTN OCONUS. The appliance supplier (i.e., LSC or MFSS supplier) should support TDM trunk groups on its MG product that can interconnect with devices in foreign country PTT networks (OCONUS) worldwide. This requirement was met for the E1 PRI (Q.931) interface.

(14) MG Support for CCS7. The MG shall support TDM trunk groups that are controlled by a separate CCA-to-SG signaling link that carries DoD CCS7 protocol. The MG shall support these TDM trunk groups, and the SG shall support DoD CCS7 signaling. This conditional requirement was not tested on the SUT.

(15) MG Support for ISDN PRI Trunks. The MG shall support ISDN PRI trunk groups that carry the U.S./National ISDN version of the ISDN PRI protocol. The SUT met all requirements for MG Support for ISDN T1 PRI Trunks.

(16) MG Support for CAS Trunks. The MG shall support CAS trunk groups that carry the U.S. version of the CAS protocol. CAS is a conditional requirement for LSCs but was tested on the SUT. The SUT had discrepancies adjudicated by DISA to be critical. The CAS interface is not certified for use.

(17) MG requirements for VoIP Internal Interfaces. The MG shall connect to the ASLAN of the appliance using the physical layer and data link layer protocols of the ASLAN. In this case, the MG shall appear to the MGC, EBC, and appliance PEIs/AEIs as a physical layer and data link layer endpoint on a LAN switch in the ASLAN. The SUT met all requirements for VoIP Internal Interfaces for PEIs.

(18) MG Echo Cancellation. The MG shall provide an EC capability with an echo path capacity (echo tail length) of at least 64 ms. The MG shall provide echo cancellation for voice, G3 Fax, and VBD modem fax calls. Each MG EC shall be equipped with an “echo canceller disabling signal” tone detector. The SUT met all requirements for MG Echo Cancellation.

(19) MG Clock Timing. The MG shall derive its clock timing from a designated T1 or PRI interface. The SUT met all MG Clock Timing requirements.

(20) MG V.150.1. When the MG uses V.150.1 inband signaling to transition between audio, FoIP, modem relay, or VBD states or modes, the MG shall continue to use the established session's protocol (e.g., decimal 17 for UDP) and port numbers so that the transition is transparent to the EBC. The V.150.1 requirement is a new UCR requirement for which the 18-month (July 2011) rule applies.

(21) MG Preservation of Call Ringing during Failure. The LSC MG, MFSS MG, and WAN SS MG shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within that MG. This requirement was not tested on the SUT. This is a new UCR requirement for which the 18-month rule applies.

i. SG Requirements.

(1) SG and CCS7 network Interactions. The SG shall support signaling connectivity to the DoD CCS7 network based on UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features, specifications for CCS7. This is a conditional requirement for LSCs and is not supported by the SUT.

(2) SG Interactions with CCA. The SG shall support a supplier-specific interface to the CCA for interactions between the SG and CCA. This is a conditional requirement for LSCs and was not tested on the SUT.

(3) SG Interworking Functions. The SG will terminate CCS7 links on its CCS7 side and transport the CCS7 call control and service control protocols (i.e., ISUP and TCAP) to the CCA. Similarly, the SG will receive CCS7 call control and service control messages from the CCA. The SG is responsible for the appropriate formatting of the messages for transmission on the CCS7 links. This is a conditional requirement for LSCs and was not tested on the SUT.

j. WWNDP Requirements.

(1) WWNDP. The precedence level and dialed number input to the PEI or AEI shall be as specified in UCR 2008, Section 5.2.3.5.1.2, Interswitch and Intraswitch Dialing. The SUT met all requirements for WWNDP for PEIs.

(2) DSN WWNDP. LSCs must support DSN WWNDP and must support mapping of DSN telephone numbers to SIP URIs, provides examples of DSN numbers using SIP URIs that use the syntax defined in RFC 3966. The SUT met all DSN WWNDP requirements for PEIs.

k. Commercial Cost Avoidance.

(1) Commercial Cost Avoidance. The LSC must use a Commercial Cost Avoidance functionality to route calls from an IP EI to a PSTN E.164 number in a manner which will minimize commercial costs associated with DSN calls. This requirement was not tested on the SUT. This represents a new feature in the UCR for which the 18-month rule applies. The vendor has until July 2011 to comply with this feature.

l. AS-SIP Based for External Devices (Voicemail, Unified Messaging and Automated Receiving Devices).

(1) AS-SIP Requirements for External Interfaces. The LSC shall support all mandatory requirements in RFC 3842. The LSC shall support all mandatory requirements in IETF Internet Draft draft-levy-sip-diversion-08.txt, Diversion Indication in SIP. No AS-SIP external devices were tested.

m. Precedence Call Diversion.

(1) Precedence Call Diversion. The AS-SIP signaling appliance shall divert ALL unanswered RTS VoIP calls above the ROUTINE precedence level to a designated RTS DN for PCD. The SUT met all precedence call diversion requirements.

n. Attendant Station Features. No attendant station was provided on the SUT at the time of test; therefore, none of the following features were tested. The Attendant Console functionality is a new UCR requirement for which the 18-month rule (July 2011) applies.

(1) Precedence and Preemption. The RTS Attendant Console shall interoperate with PBAS/ASAC.

(2) Call Display. The RTS Attendant Console shall provide a visual display of each precedence level and the calling number, for incoming direct dialed calls to the attendant, and diverted calls to the attendant.

(3) Class of Service Override. If the LSC, MFSS, or WAN SS supports assignment of a CoS to an individual EI, then this appliance and the attendant console shall give the attendant the ability to override any incoming call's calling party CoS (based on calling area or precedence) on a call-by-call basis.

(4) Busy Override and Busy Verification. The appliance and the attendant console shall give the attendant the ability to verify and override a busy line condition.

(5) Night service. The appliance and the attendant console shall have the ability to route all calls that are normally directed to the console to a separate night service deflection number.

(6) Automatic Recall of Attendant. When an attendant redirects an incoming

call to a destination station, and that station is either busy or does not answer the call within a preset time, the appliance and the attendant console shall ensure that calling party on the redirected call is recalled automatically to the console.

(7) Calls in Queue to the Attendant. The appliance and the attendant console shall have the ability to place calls (both directed to the attendant and diverted to the attendant) into a waiting queue.

o. AS-SIP Requirements.

(1) SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs. The defined for SIP EIs if they serve SIP EIs, and LSCs MUST comply with the differentiated set of requirements defined for H.323 and/or vendor-proprietary EIs if they serve H.323 and/or vendor-proprietary EIs. This requirement was not tested on the SUT. This represents a new feature in the UCR for which the 18-month rule applies. The vendor has until July 2011 to comply with this feature.

(2) SIP Session Keep-Alive Timer. The AS-SIP signaling appliances MUST support the keep-alive mechanism for SIP sessions in accordance with RFC 4028. The SUT met all keep-alive timer requirements.

(3) Session Description Protocol (SDP). A session description consists of a session-level description (details that apply to the whole session and all media streams) and optionally several media-level descriptions (details that apply to a single media stream). The LSC must support SDP in accordance with RFC 2327. The SUT met all SDP requirements.

(4) Precedence and Preemption. The LSC must meet the detailed requirements for the execution of preemption and the handling of precedence information as defined in section 5.3.4.2.10 of the UCR. The SUT met all critical precedence and preemption requirements.

(5) Video Telephony – General Rules. Video calls must meet the detailed requirements for video telephony messaging as defined in section 5.3.4.12 of the UCR. Video telephony requirements were not tested on the SUT. No video end instruments were provided on the SUT at the time of testing.

(6) Calling Services. The LSC must meet AS-SIP call flow requirements for calling services features as defined in section 5.3.4.13 of the UCR with the following exceptions: The SUT met the assured services interaction with attended transfer and call forwarding.

(7) SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances. This specification uses SIP translation for converting between ISUP signaling and AS-SIP signaling but does not use SIP encapsulation of ISUP. This requirement applies to translations between AS-SIP and CCS7. Because CCS7 is a conditional requirement for LSCs and not supported by the SUT, this requirement was not tested.

(8) Relevant Timers for the Terminating Gateway and the Originating Gateway. This requirement applies to gateways between AS-SIP and CCS7 links. Because CCS7 is a conditional requirement for LSCs and not supported by the SUT, this requirement was not tested.

(9) SIP Requirements for Interworking AS-SIP Signaling Appliances. Interworking AS-SIP signaling appliances MUST comply with UCR 2008 Section 5.3.4.7.1, AS-SIP Signaling Appliances and AS-SIP EIs, as well as the additional general requirements in UCR 2008 Section 5.3.4.16. The SUT met all requirements for interworking AS-SIP signaling appliances.

(10) Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances. Interworking AS-SIP signaling appliances MUST comply with UCR 2008, Section 5.3.4.8, SIP Session Keep-Alive Timer, as well as the additional E1 requirements listed in UCR 2008 Section 5.3.4.17. The SUT met all keep-alive timer requirements for interworking AS-SIP signaling appliances.

(11) Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances. The LSC must meet all requirements for header fields as listed in UCR 2008 section 5.3.4.18. The SUT met all requirements for precedence and preemption extensions for interworking AS-SIP signaling appliances.

(12) Supplementary Services. The LSC must meet call flow requirements as described in section 5.3.4.19 for supplementary services with the following exceptions: The SUT met the assured services interaction with attended transfer and call forwarding.

p. IPv6 Requirements.

(1) Product Requirements. This requirement was met by the SUT with both testing and vendor submission of letter of compliance (LoC). The vendor’s submitted IPv6 LoC identified partial compliance to several required Request for Comments (RFCs). Testing results did not identify any interoperability discrepancies associated with these RFC shortcomings. The DISA adjudicated the LoC discrepancies as having a minor operational impact with a vendor submitted PoAM.

q. NM. The Vendor submitted an NM LoC with noted discrepancies. Open TDRs were adjudicated by DISA to have a minor operational impact with vendor submitted PoAM.

(1) LSC Management Function. The LSC Management function supports functions for LSC FCAPS management and audit logs. This was met by the SUT with a vendor submitted LoC.

(2) VVoIP NMS Interface Requirements. The physical interface between the DISA VVoIP EMS and the network components (i.e., LSC, MFSS, EBC, CE Router) is a 10/100-Mbps Ethernet interface. The interface will work in either of the two following modes using auto-negotiation: IEEE, Ethernet Standard 802.3, 1993; or IEEE, Fast Ethernet Standard 802.3u, 1995. This was met by the SUT with a vendor submitted LoC.

(3) General Management Requirements. The LSC components shall each have an individual pair of Ethernet interfaces for management purposes, even in cases where the MFSS or LSC component contains multiple physical devices. This was met by the SUT with a vendor submitted LoC with the exception that the SUT supports only voice management. The DISA adjudicated this discrepancy as having a minor operational impact with vendor submitted PoAM.

(4) Requirement for FCAPS Management. The LSC must meet all general requirements for the FCAPS management functional areas as defined in UCR 2008 Section 5.3.2.17. This was met by the SUT with a vendor submitted LoC.

(5) NM requirements of Appliance Functions. The LSC must meet all management requirements for ASAC, CCA, SG, and MG functions as defined in UCR 2008 Section 5.3.2.18. The SUT does not provide the directionalization function for incoming and outgoing ASAC counts. The feature has been made “Conditional” in UCR 2010. The DISA adjudicated this discrepancy as having a minor operational impact with vendor submitted PoAM.

(6) Accounting Management. Accounting management identifies a set of events during which call detail information is collected. These events are call connect, call attempt, and call disconnect. When these events are detected, specific call data will be provided by the network appliances that were involved in the event. The SUT does not comply with the objective requirement for Record Format.

11.3 Information Assurance. The IA report is published in a separate report, Reference (e).

11.4 Other. None.

12. TEST AND ANALYSIS REPORT. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System 2-7 Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi. All associated data is available on the Defense Information Systems Agency Unified Capability Coordination Office (UCCO) website located at http://www.disa.mil/ucco/.

SYSTEM FUNCTIONAL AND CAPABILITY REQUIREMENTS

The local session controllers have required and conditional features and capabilities that are established by the Unified Capabilities Requirements (UCR). The System Under Test (SUT) need not provide conditional requirements. If they are provided, they must function according to the specified requirements. The detailed Functional requirements (FR) and Capability Requirements for Internet Protocol Call Control products (Multi-Function SoftSwitch (MFSS), Local Session Controller (LSC), and Wide Area Network SoftSwitch (WAN SS) are listed in Table 3-1. Detailed Information Assurance (IA) requirements are included in Reference (e) and are not listed below.

Table 3-1. LSC Products Capability/Functional Requirements Table

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
1	As part of the session setup process, the LSC controls what DSCP to use in the subsequent session media stream packets. (two sub requirements)	5.3.2.2.1.4	Y	Y	
2	The SUT must provide the following features: Precedence Call Waiting, Call Forwarding, Call Transfer, Call Hold, Three-Way Calling, Hotline Service, and Calling Party and Called Party ID.	Table 5.3.2.2-1	Y	Y	
3	Calls to a DN that does not have any CF feature activated shall be delivered to the DN EI IAW the MLPP procedures specified in UCR 2008, Section 5.2.2 Multilevel Precedence and Preemption	5.3.2.2.2.1.1	Y	Y	
4	Call forwarding, when activated on a line DN, shall allow any terminating call at a ROUTINE DSN precedence level, to be completed to the designated destination (IAW the call forward options activated), and shall comply with the requirements as stated in Telcordia Technologies GR-217-CORE, GR-580-CORE, and GR-586-CORE.	5.3.2.2.2.1.1	Y	Y	
5	Calls to 911 shall be preempted in accordance with assured service priority rules specified in UCR 2008, Section 5.2.2, Multilevel Precedence and Preemption.	5.3.2.2.2.2.1	Y	Y	
6	The Tracing of Terminating Calls feature identifies the calling number on intraoffice and interoffice calls terminating to a specified DN. When this feature is activated, the originating DN, the terminating DN, and the time and date are printed out for each call to the specified line.	5.3.2.2.2.2.2	Y	Y	
7	The Outgoing Call Tracing feature allows the tracing of nuisance calls to a specified DN suspected of originating from a given local office. The tracing is activated when the specified DN is entered. A printout of the originating DN, and the time and date, are generated for every call to the specified DN.	5.3.2.2.2.2.3	Y	Y	
8	The Tracing of a Call in Progress feature identifies the originating DN for a call in progress. Authorized personnel entering a request that includes the specific terminating DN involved in the call activate the feature.	5.3.2.2.2.2.4	Y	Y	
9	The Tandem Call Trace feature identifies the incoming trunk of a tandem call to a specified office DN. The feature is activated by entering the specified distant office DN for a tandem call trace. A printout of the incoming trunk number and terminating DN, and the time and date, is generated for every call to the specified DN.	5.3.2.2.2.2.5	Y	Y	
10	One voice session budget unit shall be equivalent to 110 kilobits per second (kbps) of access circuit bandwidth independent of the PEI or AEI codec used. This includes ITU-T Recommendation G.711 encoding rate plus Internet Protocol Version 6 (IPv6) packet overhead plus ASLAN Ethernet overhead. IPv6 overhead, not IPv4 overhead, is used to determine bandwidth equivalents here.	5.3.2.2.2.3.1	Y	Y	
11	If the MFSS's count of an IPC is greater than or equal to the corresponding IPB, and it receives an INVITE request for a precedence session, the MFSS shall preempt a lower priority session (if such a session exists), and then proceed with processing the higher precedence session connect request.	5.3.2.2.2.3.1.2	Y		

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
12	If the MFSS receives a CCA-ID for which there is no entry in ASAC budget table, the SS will reject the session and generate a alarm for the EMS.	5.3.2.2.2.3.1.2	Y		
13	If necessary, the MFSS will preempt for a session request that is at precedence level FLASH OVERRIDE or FLASH and the counts equal the budgets.	5.3.2.2.2.3.2	Y		
14	Registration and authentication between NEs shall follow the requirements set forth in UCR 2008, Section 5.4, Information Assurance Requirements.	5.3.2.3.1	Y	Y	
15	The LSCs shall be registered to a primary and backup MFSS. In case of failure of the primary MFSS, the LSC will default to the backup MFSS.	5.3.2.3.2	Y	Y	
16	The LSC shall send an OPTIONS request with a Request-URI identifying the primary SS (the Request-URI does not have a userinfo part) on a configurable periodic time interval (default equals 45 seconds; minimum time interval equals 35 seconds). (4 sub requirements)	5.3.2.3.2.2		Y	
17	When a properly functioning primary SS receives the OPTIONS request from a served LSC, the primary SS shall respond with a 200 OK response that includes the Accept header and the Supported header.	5.3.2.3.2.2	Y		Y
18	When the LSC sends a defined configurable number of successive OPTIONS requests (default equals 2) for which there either is no response or the response is a 408 (Request Time-Out), 503 (Service Unavailable), or 504 (Server Time-Out) response, then it must failover to the secondary SS. (3 sub requirements)	5.3.2.3.2.3		Y	
19	If the LSC receives a 200 OK response to an OPTIONS request from the primary SS before the configurable number of successive failures to the OPTIONS requests (default equals 2) has been reached, then no action is taken to failover to the secondary SS.	5.3.2.3.2.3		Y	
20	Upon failover, the LSC will send OPTIONS requests to the primary SS at a failback configurable periodic time interval (default equals 60 seconds; minimum time interval equals 35 seconds). (4 subrequirements)	5.3.2.3.2.4		Y	
21	Each SS shall send an OPTIONS request to every other SS on a “standard” configurable periodic time interval (default equals 45 seconds; minimum time interval equals 35 seconds).	5.3.2.3.2.5	Y		Y
22	Whenever an originating SS sends an INVITE request to another SS and receives either a 408 (Request Time-Out), 503 (Service Unavailable), or 504 (Server Time-Out) response and the originating SS is not already awaiting a response to a pending OPTIONS request to the other SS, then the originating SS shall send an OPTIONS request with a Request-URI identifying the SS.	5.3.2.3.2.5	Y		Y
23	When a properly functioning SS receives the OPTIONS request, the SS shall respond with a 200 OK response that includes the Accept header and the Supported header.	5.3.2.3.2.5	Y		Y
24	Each MFSS (SS) shall be configured with knowledge of each pair of SSs that act as backups for each other. (7 subrequirements)	5.3.2.3.2.6	Y		Y
25	Upon failover, the SS will send OPTIONS requests to the failed SS at a “failback” configurable periodic time interval (default equals 60 seconds; minimum time interval equals 35 seconds). (5 subrequirements)	5.3.2.3.2.7	Y		Y
26	The Assured Services subsystem shall have a hardware/software availability of 0.99999 (nonavailability of no more than 5 minutes per year).	5.3.2.5.2.1	Y	Y	
27	The performance parameters associated with the ASLAN, MFSS, and LSC, when combined, shall meet the following maximum downtime requirements: • IP (10/100 Ethernet) network links – 35 minutes/year • IP subscriber – 12 minutes/year	5.3.2.5.2.2	Y	Y	
28	For these VoIP devices, the voice quality shall have a MOS of 4.0 (R-Factor equals 80) or better, as measured in accordance with the E-Model. Additionally, these devices shall not lose two or more consecutive packets in a minute and shall not lose more than seven	5.3.2.5.4	Y	Y	

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
	voice packets (excluding signaling packets) in a 5-minute period.				
29	An IP voice instrument shall be designed in accordance with the acquiring activity requirements, but the following capabilities are specifically required as indicated: • **[Objective]** DoD Common Access Card (CAC) reader • **[Required]** Display calling number • **[Required]** Display precedence level of the session • **[Required]** Support for Dynamic Host Configuration Protocol (DHCP).	5.3.2.6.1	Y	Y	
30	Tones and announcements, as required in UCR 2008, Sections 5.2.4.5.2, DSN Information Signals, and Section 5.2.2.1.3, Announcements, shall be supported, except for the loss of C2 announcement.	5.3.2.6.1.1	Y	Y	
31	The product shall support the origination and termination of a voice session using the following codecs: • ITU-T Recommendation G.711, to include both the µ-law and A-law algorithms • ITU-T Recommendation G.723.1 • ITU-T Recommendation G.729 or G.729A • ITU-T Recommendation G.722.1	5.3.2.6.1.2	Y	Y	
32	Voice over IP PEIs or AEIs (i.e., handset, headset, and hands-free types) shall comply with TIA-810-B, November 3, 2006.	5.3.2.6.1.3	Y	Y	
33	For Fixed-to-Fixed calls, the product shall use 20 ms as the default voice sample length, and as the basis for the voice payload packet size.	5.3.2.6.1.4	Y	Y	
34	The PEI or AEI shall be capable of authenticating itself to its associated LSC and vice versa.	5.3.2.6.1.5 5.3.2.6.2.3	Y	Y	
35	Analog instruments, including secure analog EIs, analog facsimile EIs, and analog modem EIs, shall be supported by the LSC either by a TA or an Integrated Access Device (IAD) connected to an Ethernet port.	5.3.2.6.1.6	Y	Y	
36	The LSC shall meet all the requirements for PBAS/ASAC, as appropriate for VoIP and Video over IP services, as specified in UCR 2008, Section 5.2.2, Multilevel Precedence and Preemption.	5.3.2.7.2.1		Y	
37	The LSC shall support CND, as specified in UCR 2008, Section 5.2.3.5.1.8.2, Calling Number Delivery.	5.3.2.7.2.2		Y	
38	The LSC must provide signaling on the line side for local intra-enclave subscriber-to-subscriber calls, and trunk-side signaling for calls between an external enclave and a local subscriber.	5.3.2.7.2.3		Y	
39	In the event that a total loss of connectivity to the DISN WAN occurs, the LSC shall provide the following functions: • Completion of local (intra-enclave) calls • Routing of calls to the PSTN using a local MG (PRI or CAS as required by the local interface) • User look-up of local directory information	5.3.2.7.2.4		Y	
40	The LSC Management function supports functions for LSC FCAPS management and audit logs. Collectively, these functions are called FCAPS Management and Audit Logs.	5.3.2.7.2.6		Y	
41	The LSC Transport Interface functions provide interface and connectivity functions with the ASLAN and its IP packet transport network.	5.3.2.7.2.7		Y	
42	The LSC shall provide an interface to the DISA NMS. The interface consists of a 10/100-Mbps Ethernet connection	5.3.2.7.2.8		Y	
43	Periodically, the LSC shall verify the status of its registered and authenticated IP EIs, including operator (dial service attendant) consoles. The verification interval shall be configurable with the default set at 5 minutes.	5.3.2.7.2.10		Y	
44	Line-side custom features must not interfere with the Assured Services requirements.	5.3.2.7.2.11		Y	

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
45	During the call establishment process, the product shall be capable of preventing or detecting and stopping hair-pin routing loops over ANSI T1.619a and commercial PRI trunk groups (i.e., T1 PRI and E1 PRI) between a legacy switch (e.g., TDM EO) and an LSC	5.3.2.7.3		Y	
46	When the AS-SIP TDM Gateway receives a call request over an ISDN MLPP PRI then the AS-SIP TDM Gateway MUST map the telephony numbers received from the Q.931 SETUP message to SIP URIs	5.3.2.7.4.3.3	Y	Y	
47	The AS-SIP TDM Gateway MG MUST support the ITU-T Recommendation G.711 (μ-law and A-law) audio codec.	5.3.2.7.4.3.4	Y	Y	
48	The AS-SIP TDM Gateway MG MUST support RFC 4040 and the AS-SIP TDM Gateway MUST support the signaling for establishing the 64kbps unrestricted bearer per Section 5.3.4.7.7, 64 kbps Transparent Calls (Clear Channel).	5.3.2.7.4.3.4	Y	Y	
49	The AS-SIP TDM Gateway MG MUST support T.38 Fax Relay	5.3.2.7.4.3.4	Y	Y	
50	The AS-SIP TDM Gateway MG MUST support the SCIP-216 subset of V.150.1 Modem Relay (see Section 5.3.2.21.2, RTS SCIP Gateway Requirements) and the AS-SIP TDM Gateway MUST support the AS-SIP signaling requirements in support of modem relay	5.3.2.7.4.3.4	Y	Y	
51	The AS-SIP TDM Gateway MUST satisfy the Information Assurance requirements in Section 5.4 Information Assurance for a media gateway.	5.3.2.7.4.3.5	Y	Y	
52	The AS-SIP TDM Gateway MUST provide an interface to the DISA NMS. The interface MUST consist of a 10/100-Mbps Ethernet connection	5.3.2.7.4.3.9	Y	Y	
53	The AS-SIP IP Gateway MUST implement call count thresholds for voice sessions and for video sessions in order to perform Session Admission Control (SAC).	5.3.2.7.5.1.1	Y	Y	
54	The CCA IWF must support AS-SIP and ISDN PRI protocols.	5.3.2.9.2.1	Y	Y	
55	The MGC within the CCA must control all MGs within the LSC or MFSS, support DoD ISDN trunks, control all signaling and media streams on each trunk within each MG, and accept IP-encapsulated signaling streams from an SG or MG.	5.3.2.9.2.2	Y	Y	
56	The CCA shall be responsible for controlling all the SGs within the MFSS and LSC.	5.3.2.9.2.3	Y	C	
57	The CCA shall be responsible for controlling each signaling link within each SG within the MFSS or LSC.	5.3.2.9.2.3	Y	C	
58	The CCA shall be responsible for controlling the DoD CCS7 signaling stream(s) within each signaling link within each SG.	5.3.2.9.2.3	Y	C	
59	Within the network appliance (i.e., MFSS and LSC), the CCA shall use either an IETF-standard set of CCS7-over-IP protocols, or a supplier-proprietary protocol to accomplish the above SG, signaling link, and signaling stream controls.	5.3.2.9.2.3	Y	C	
60	The CCA IWF shall support the AS-SIP protocol consistent with the detailed AS-SIP protocol requirements in Section 5.3.4, AS-SIP Requirements.	5.3.2.9.5.1	Y	Y	
61	The CCA IWF shall use the AS-SIP protocol on LSC-MFSS and MFSS-MFSS sessions.	5.3.2.9.5.1	Y	Y	
62	When the CCA IWF uses the AS-SIP protocol over the Access Segment between the EBC and the DISN WAN, or over the DISN WAN itself, the CCA IWF shall secure the AS-SIP protocol using TLS.	5.3.2.9.5.1	Y	Y	
63	The CCA IWF shall support the U.S./National ISDN version of the ISDN PRI protocol.	5.3.2.9.5.3	Y	Y	
64	The CCA IWF shall support reception of ISDN PRI messages from the MG and transmission of ISDN PRI messages to the MG.	5.3.2.9.5.3	Y	Y	
65	The CCA IWF shall be able to determine the ISDN PRI (and its D-Channel signaling link) that an incoming PRI message was received on, when processing an incoming PRI message from the MG.	5.3.2.9.5.3	Y	Y	

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
66	The CCA IWF shall be able to identify the ISDN PRI (and its D-Channel signaling link) that an outgoing PRI message will be sent on, when generating an outgoing PRI message to the MG.	5.3.2.9.5.3	Y	Y	
67	The CCA IWF shall be able to support multiple ISDN PRIs (and their D-Channel signaling links) at the MG, where each PRI is connected to a different PRI end point.	5.3.2.9.5.3	Y	Y	
68	The CCA IWF shall be able to differentiate between the individual ISDN PRIs (and their D-Channel signaling links) at the MG.	5.3.2.9.5.3	Y	Y	
69	The CCA IWF shall support the full set of ISDN MLPP requirements in ANSI T1.619 and ANSI T1.619a.	5.3.2.9.5.3	Y	Y	
70	The CCA IWF shall not support any of the ISDN MLPP requirements in ANSI T1.619 and ANSI T1.619a, on ISDN PRIs to TDM PBXs and switches in the U.S. PSTN.	5.3.2.9.5.3	Y	Y	
71	On ISDN PRIs from the CCA/MG to TDM PBXs and switches in allied and coalition partners (where those networks support U.S. "National ISDN" PRI), the CCA IWF shall support a DoD-user-configurable per-PRI option that allows the PRI to support or not support the ANSI T1.619/619a PRI MLPP feature on calls to and from that PRI.	5.3.2.9.5.3	Y	Y	
72	The CCA IWF shall be able to associate individual PRI configuration data with each individual PRI served by the MG and the CCA. The CCA IWF shall not require groups of PRIs served by the MG and the CCA to share "common" PRI configuration data.	5.3.2.9.5.3	Y	Y	
73	The CCA IWF shall support supplier-proprietary Voice and Video EIs and their associated proprietary EI signaling protocols.	5.3.2.9.5.5	Y	Y	
74	The CCA in the LSC, MFSS, and WAN SS shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within the CCA.	5.3.2.9.6	Y	Y	
75	The MFSS CCA shall be able to support MG connections between the SS side of the MFSS and the EO and Tandem functions on the TDM side of the MFSS.	5.3.2.10.1	Y		
76	The CCA shall support assignment of the following items to itself: • Only one CCA IP address (this one IP address may be implemented in the CCA as either a single logical IP address or a single physical IP address), • A CCA Fully Qualified Domain Name (FQDN) that maps to that IP address, and • A CCA SIP URI that uses that CCA FQDN as its domain name, and maps to the "SIP B2BUA" function within the CCA itself.	5.3.2.10.3	Y	Y	
77	The CCA shall support assignment of the following items to each SIP and AS-SIP PEI and AEI on the Appliance LAN: • Only one PEI or AEI IP address, • A PEI or AEI FQDN that maps to that IP address, and • A PEI or AEI SIP URI that uses that PEI or AEI FQDN as its domain name, and maps to the "SIP User Agent" function within the PEI or AEI.	5.3.2.10.3	Y	Y	
78	The CCA shall support assignment of the following items to each MG on the Appliance LAN: • Only one MG IP address (this one IP address may be implemented in the MG as either a single logical IP address or a single physical IP address), • An MG FQDN that maps to that IP address, and • An MG SIP URI that uses that MG FQDN as its domain name, and maps to the "UC Signaling and Media End Point" function within the MG.	5.3.2.10.3	Y	Y	
79	The CCA shall support assignment of the following items to each SG on the Appliance LAN: • Only one SG IP address (this one IP address may be implemented in the SG as either a single logical IP address or a single physical IP address), • An SG FQDN that maps to that IP address, and • An SG SIP URI that uses that SG FQDN as its domain name, and	5.3.2.10.3	Y	C	

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
	maps to the "UC Signaling End Point" function within the SG				
80	The CCA shall support assignment of the following items to the EBC: • Only one EBC IP address (this one IP address may be implemented in the EBC as either a single logical IP address or a single physical IP address), • An EBC FQDN that maps to that IP address, and • An EBC SIP URI that uses that EBC FQDN as its domain name, and maps to the "SIP B2BUA" function within the EBC.	5.3.2.10.3	Y	Y	
81	When directing VoIP sessions to other network appliances providing voice and video services across the DISN, the CCA shall direct these VoIP sessions to the EBC, so that the EBC can process them before directing them to the network appliances on the DISN WAN.	5.3.2.10.4	Y	Y	
82	When accepting VoIP sessions from other network appliances on the DISN, the CCA shall accept these VoIP sessions from the EBC, because the EBC relays them from the network appliances on the DISN WAN.	5.3.2.10.4	Y	Y	
83	The LSC and MFSS CCA shall meet all the requirements in Section 5.3.2.2.2.3, ASAC – Open Loop. The LSC and MFSS CCA shall meet all the requirements in Section 5.3.4.10, Precedence and Preemption. The LSC and MFSS CCA shall meet all the requirements in Section 5.3.4.11, Policing of Call Count Thresholds.	5.3.2.10.5	Y	Y	
84	The CCA shall generate a redirecting number each time it forwards a VoIP or Video session request as part of a Call Forwarding feature.	5.3.2.10.6	Y	Y	
85	It is expected that all Assured Services products, such as LSCs and MFSSs, will support vendor-proprietary VVoIP features and capabilities, in addition to supporting the required VVoIP features and capabilities that are listed.	5.3.2.10.6	Y	Y	
86	The CCA shall relay received SIP and TLS authentication credentials and encryption key information from sending end systems (i.e., users, PEIs, AEIs, and EBCs) to the Information Assurance function to support the Information Assurance function's user, PEI, AEI, and EBC authentication capabilities, and its PEI, AEI, and EBC signaling stream encryption capabilities.	5.3.2.10.7	Y	Y	
87	The CCA MGC shall relay received H.248 and IPSec (or proprietary-protocol-equivalent) authentication credentials and encryption key information from sending end systems (i.e., MGs and SGs) to the Information Assurance function to support the Information Assurance function's MG and SG authentication capabilities, and its MG and SG signaling stream encryption capabilities.	5.3.2.10.7	Y	Y	
88	The CCA shall relay authentication credentials received in a SIP or AS-SIP REGISTER message from an PEI, AEI, or EBC to the Information Assurance function.	5.3.2.10.7	Y	Y	
89	The CCA shall relay TLS encryption key information received from a PEI or AEI to the Information Assurance function so the Information Assurance function can verify that this encryption key information can be used on the signaling streams for Voice or Video sessions to/from that PEI or AEI.	5.3.2.10.7	Y	Y	
90	The CCA shall relay TLS encryption key information received from an EBC to the Information Assurance function so the Information Assurance function can verify that this encryption key information can be used on the signaling streams for the Voice or Video sessions to/from that EBC.	5.3.2.10.7	Y	Y	
91	The CCA within the appliance shall support all Information Assurance Appliance requirements in Section 5.4, Information Assurance Requirements, which involve the appliance's SCS functions and the appliance's MGC.	5.3.2.10.7	Y	Y	
92	The CCA shall support supplier-proprietary Voice and Video EIs, using EI-CCA protocols that are proprietary to the LSC or MFSS supplier.	5.3.2.10.10	Y	Y	
93	When the CCA IWF supports AS-SIP Voice and Video AEIs, the IWF shall support these AEIs using the set of AS-SIP protocol requirements in Section 5.3.2.22, Generic AS-SIP End Instrument and Video Codec Requirements, and Section 5.3.4, AS-SIP Requirements.	5.3.2.10.10	Y	Y	

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
94	The Appliance CCA (i.e., LSC or MFSS) shall support both assured Voice and Video services. The CCA shall support both assured Voice and assured Video sessions, and shall support these sessions from both VoIP EIs and Video EIs, as described in UCR 2008, Section 5.3.2.10.10, CCA Interactions with End Instrument(s).	5.3.2.10.11	Y	Y	
95	The Appliance CCA shall support common procedures and protocol for VoIP and Video session control.	5.3.2.10.11	Y	Y	
96	The Appliance CCA shall support common procedures and protocol for feature control, for the features and capabilities given in Table 5.3.2.2-1, Assured Services Product Features and Capabilities.	5.3.2.10.11	Y	Y	
97	On calls to and from Proprietary VoIP and Proprietary Video EIs, the CCA shall use the appropriate parameters within the appliance supplier's Proprietary protocol messages to differentiate Proprietary VoIP sessions from Proprietary Video sessions.	5.3.2.10.11	Y	Y	
98	When AS-SIP EIs are supported on calls to and from AS-SIP EIs, the CCA shall use the SDP message bodies in AS-SIP INVITE, UPDATE, REFER, and ACK messages, as well as the SDP message bodies in AS-SIP 200 OK responses and earlier 1xx provisional responses, to differentiate AS-SIP Voice sessions from AS-SIP Video sessions.	5.3.2.10.11	Y	Y	
99	The CCA shall track VoIP sessions against corresponding Appliance VoIP budgets, and shall separately track Video sessions against corresponding Video budgets. The CCA shall maintain the Appliance's VoIP budgets separate from the Appliance's Video budget.	5.3.2.10.11	Y	Y	
100	As part of LSC-Level ASAC and WAN-Level ASAC Policing, the CCA shall support PBAS/ASAC for both VoIP sessions and Video sessions.	5.3.2.10.11	Y	Y	
101	The CCA shall allow an individual EI to support both VoIP and Video sessions. The CCA shall allow an individual EI to have both VoIP and Video sessions active at the same time.	5.3.2.10.11	Y	Y	
102	The CCA shall support the routing of both VoIP and Video session requests from LSCs to MFSSs, from MFSSs to LSCs, and from MFSSs to MFSSs, using AS-SIP. The CCA shall direct outgoing VoIP and Video session requests to EBCs, and shall accept incoming VoIP and Video session requests from EBCs, consistent with this LSC-to-MFSS routing, MFSS-to-LSC routing, and MFSS-to-MFSS routing.	5.3.2.10.11	Y	Y	
103	The CCA shall support the ability to remove VoIP and Video sessions and session requests from the media server so the CCA can continue with necessary session processing once the media server has completed its functions.	5.3.2.10.12	Y	Y	
104	The MG supports interconnection of VoIP, FoIP, and MoIP media streams with the following LSC functions and end-user devices: a. The LSC media server, which provides tones and announcements for LSC calls and LSC features b. AS-SIP VoIP, FoIP, and MoIP AEIs on the LSC	5.3.2.12.3.1		Y	
105	The MFSS MG shall be able to support MG trunk groups (referred to as internal MG connections) that either interconnect the SS (VoIP) side of the MFSS with the EO or Tandem functions on the TDM side of the MFSS.	5.3.2.12.3.2.1	Y		
106	On incoming call requests to a TDM trunk group, where the CCA/MGC applies a CAC Call Denial treatment to that call request, the MG shall connect the TDM called party on the incoming call request to the appropriate CAC Call Denial tone or announcement when instructed to do so by the MGC.	5.3.2.12.4.1.1	Y	Y	
107	On incoming calls or call requests to a TDM trunk group, where the CCA/MGC applies an ASAC Call Preemption treatment to that call or call request, the MG shall connect the TDM calling party on the incoming call or call request to the appropriate ASAC Call Preemption tone or announcement when instructed to do so by the MGC.	5.3.2.12.4.1.1	Y	Y	
108	On outgoing calls or call requests from a TDM trunk group, where the CCA/MGC applies an ASAC Call Preemption treatment to that call or call request, the MG shall connect the TDM called party on the outgoing call or call request to the appropriate ASAC Call Preemption tone or announcement when instructed to do so by the MGC.	5.3.2.12.4.1.1	Y	Y	
109	Each MG within an appliance shall support all the appliance requirements in Section 5.4, Information Assurance Requirements, that involve an Appliance MG.	5.3.2.12.4.2	Y	Y	

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
110	When instructed to do so by the MGC, the MG shall direct TDM calls and call requests to the media server.	5.3.2.12.4.3	Y	Y	
111	Since each Appliance MG is an IP endpoint on the Appliance LAN, each MG shall support assignment of the following items to itself: • Only one MG IP address (This one IP address may be implemented in the CCA as either a single logical IP address or a single physical IP address.) • An MG FQDN that maps to that IP address • An MG SIP URI that uses that MG FQDN as its domain name, and maps to a "SIP User Agent" function within the MG.	5.3.2.12.4.4	Y	Y	
112	The MG shall interact with the Transport Interface functions in the appliances when the MG uses the native LAN protocols, IP, and UDP to exchange SRTP media streams with PEIs, AEIs, other MGs, and the EBC over the Appliance LAN	5.3.2.12.4.4	Y	Y	
113	When sending VoIP media streams to PEIs or AEIs and MGs served by other network appliances, the MG shall direct these VoIP media streams to the EBC so the EBC can process them before sending them on to the remote PEIs or AEIs and MGs via the DISN WAN.	5.3.2.12.4.5	Y	Y	
114	When accepting VoIP media streams from PEIs or AEIs and MGs served by other network appliances, the MG shall accept these VoIP media streams from the appliance EBC, because the EBC relays them from the DISN WAN and the remote PEIs or AEIs and MGs on the DISN WAN. The MG shall recognize and act on the network-level IP addresses of the remote PEIs or AEIs and MGs, when accepting the VoIP sessions through the EBC from the DISN WAN and the remote PEIs or AEIs and MGs.	5.3.2.12.4.5	Y	Y	
115	The MG shall support the exchange of VoIP media streams with the following voice PEIs and AEIs both on the local appliance and on remote network appliances: a. Supplier-proprietary voice PEIs b. Voice SIP EIs, when the appliance supplier supports these EIs c. Voice H.323 EIs, when the appliance supplier supports these EIs d. Voice AS-SIP AEIs	5.3.2.12.4.8	Y	Y	
116	The MG shall support the operation of the following features for VoIP and Video end users, consistent with the operation of this feature on analog and ISDN lines in DoD TDM switches today: • Call Hold • Music on Hold • Call Waiting • Precedence Call Waiting • Call Forwarding Variable • Call Forwarding Busy Line • Call Forwarding No Answer • Call Transfer • Three-Way Calling • Hotline Service • Calling Party and Called Party ID (number only) • Call Pickup	5.3.2.12.4.9	Y	Y	
117	Each appliance MG shall support TDM trunk groups that can interconnect with the following NEs in DoD networks, in the United States and worldwide: • PBXs • SMEOs • EOs • MFSs Media Gateway support for these TDM trunk groups shall be identical to the support for these trunk groups in DoD TDM PBXs, EOs, Tandem switches, and MFSs today, as specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features.	5.3.2.12.5	Y	Y	
118	Each appliance MG shall support TDM trunk groups that can interconnect with NEs in the PSTN in the United States, including CONUS, Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Media Gateway support for these TDM trunk groups to the U.S. PSTN shall be identical to the support for these trunk groups in DoD TDM PBXs, EOs, Tandem Switches, and MFSs today, as specified in UCR	5.3.2.12.7	Y	Y	

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
	2008, Section 5.2, Circuit-Switched Capabilities and Features.				
119	The MG shall support foreign country ISDN PRI, where the MG handles both the media channels and the signaling channel: 1. For interconnection with a foreign country PSTN using foreign country ISDN PRI, from the country where the DoD user's B/P/C/S is located. 2. Support for ETSI PRI is required on LSC trunk groups when the LSC is used in OCONUS ETSI-compliant countries. 3. Support for ETSI PRI is required on MFSS trunk groups when the MFSS is used in OCONUS ETSI-compliant countries. 4. Support for MLPP using ISDN PRI is not required on the above trunk groups.	5.3.2.12.8	Y	Y	
120	The MG shall support ISDN PRI trunk groups that carry the U.S./National ISDN version of the ISDN PRI protocol. The MG shall support these U.S. PRI trunk groups conformant with the detailed U.S. ISDN PRI requirements.	5.3.2.12.10	Y	Y	
121	The MG shall support multiple U.S. PRI trunk groups based on the needs of the DoD user deploying the appliance. The MG shall allow each U.S. PRI trunk group at the MG to connect to: TDM EO and tandem components of the local MFSS; a different U.S. PSTN TDM NE (e.g., PBX, TDM switch); a different DoD TDM NE (e.g., PBX, TDM switch); or a different DoD IP NE (e.g., LSC, MFSS), based on the interconnection needs of the DoD user.	5.3.2.12.10	Y	Y	
122	The MG shall support reception of ISDN PRI messages from the CCA MGC and transmission of ISDN PRI messages to the CCA MGC.	5.3.2.12.10	Y	Y	
123	The MG shall connect to the ASLAN of the appliance using the physical layer and data link layer protocols of the ASLAN. In this case, the MG shall appear to the MGC, EBC, and appliance PEIs/AEIs as a physical layer and data link layer endpoint on a LAN switch in the ASLAN.	5.3.2.12.12.1	Y	Y	
124	The MG shall connect to the ASLAN of the appliance using the IP as a Network Layer Protocol. In this case, the MG shall appear to the MGC, EBC, and appliance PEIs/AEIs as an IP endpoint on an IP router on the ASLAN	5.3.2.12.12.2	Y	Y	
125	The MG shall support IPv4 as a Network Layer Protocol. The MG shall also support IPv6 as a Network Layer Protocol.	5.3.2.12.12.2	Y	Y	
126	Conformant with Section 5.3.5, IPv6 Requirements, the MG shall support dual IPv4 and IPv6 stacks (i.e., support both IPv4 and IPv6 in the same IP end point) as described in RFC 4213.	5.3.2.12.12.2	Y	Y	
127	The MG shall support exchange of VoIP media streams with appliance PEIs/AEIs, other appliance MGs, and the appliance EBC (and through the appliance EBC, with other PEIs/AEIs and MGs on other network appliances) using the following IETF-defined Media Transfer Protocols: • SRTP, conformant with RFC 3711 • SRTCP, conformant with RFC 3711	5.3.2.12.12.4	Y	Y	
128	The MG shall secure all VoIP media streams exchanged with appliance PEIs/AEIs, other appliance MGs, and the appliance EBC (and through the EBC, with PEIs/AEIs and MGs on other network appliances) using SRTP and SRTCP.	5.3.2.12.12.4	Y	Y	
129	The MG shall use UDP as the underlying Transport Layer Protocol, and IP as the underlying Network Layer Protocol, when SRTP is used for media stream exchange.	5.3.2.12.12.4	Y	Y	
130	When the VoIP signaling streams contain supplier-proprietary protocol messages instead of H.248 or ISDN PRI messages, the MG shall secure the proprietary protocol message exchange with the MGC using mechanisms that are as strong as, or stronger than, the use of IPSec to secure H.248 and PRI message exchange.	5.3.2.12.12.5	Y	Y	
131	The MG shall support TDM voice streams using the following: • ITU-T 64 kbps G.711 µ-law PCM over digital trunks • ITU-T 64 kbps G.711 A-law PCM over digital trunks • North American 56 kbps G.711 µ-law PCM over digital trunks • North American analog voice transmission over analog trunks on TDM trunk groups on the TDM side of the MG	5.3.2.12.12.6.5	Y	Y	

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
132	The MG shall convert between North American 56 kbps G.711 µ-law PCM and ITU-T 64 kbps G.711 µ-law PCM in cases where North American 56 kbps TDM voice trunks are used on the TDM side of the MG.	5.3.2.12.12.6.5	Y	Y	
133	The MG shall convert between North American analog voice transmission and ITU-T 64 kbps G.711 µ-law PCM in cases where North American analog voice trunks are used on the TDM side of the MG.	5.3.2.12.12.6.5	Y	Y	
134	The MG shall support uncompressed, packetized VoIP streams using ITU-T Recommendation G.711 µ-law PCM and ITU-T Recommendation G.711 A-law PCM (ITU-T Recommendation G.711, November 1998, plus Appendix I, September 1999, and Appendix II, September 2000) over the IP network on the VoIP side of the MG.	5.3.2.12.12.6.5.1	Y	Y	
135	The MG shall packetize/depacketize G.711 media streams received or sent between its TDM side and its VoIP side.	5.3.2.12.12.6.5.1	Y	Y	
136	The MG shall transport each packetized G.711 VoIP stream to and from the destination local PEI, local AEI, local MG, remote PEI (via an EBC), remote AEI (via an EBC), or remote MG (via an EBC) using SRTP, UDP, and IP protocol layers on the VoIP side of the MG.	5.3.2.12.12.6.5.1	Y	Y	
137	The MG shall support the use of uncompressed, packetized G.711 µ-law and A-law VoIP media streams for both Fixed and Deployable applications.	5.3.2.12.12.6.5.1	Y	Y	
138	The MG shall provide an EC capability with an echo path capacity (echo tail length) of at least 64 ms.	5.3.2.12.13.2.2	Y	Y	
139	The MG shall provide echo cancellation for voice, G3 Fax, and VBD modem fax calls.	5.3.2.12.13.2.2	Y	Y	
140	Each MG EC shall be equipped with an “echo canceller disabling signal” tone detector. This tone detector shall detect and respond to an in-band EC disabling signal from an end user’s G3 Fax or VBD modem device. The EC disabling signal detected shall consist of a 2100-Hz tone with periodic phase reversals inserted in that tone.	5.3.2.12.13.2.2	Y	Y	
141	The MG tone detector/EC disabler shall detect the “echo canceller disabling signal” and disable the MG EC when, and only when, that signal is present for G3 Fax or VBD modem.	5.3.2.12.13.2.2	Y	Y	
142	The MG shall derive its clock timing from a designated T1 or PRI interface.	5.3.2.12.14	Y	Y	
143	The MGC within the CCA shall be responsible for controlling all the MGs within the LSC or MFSS.	5.3.2.12.15	Y	Y	
144	The MGC within the CCA shall be responsible for controlling all the trunks (i.e., DoD CCS7, PRI, or CAS) within each MG within the LSC or MFSS.	5.3.2.12.15	Y	Y	
145	The MGC within the CCA shall be responsible for controlling all media streams on each trunk within each MG.	5.3.2.12.15	Y	Y	
146	The MGC within the CCA shall accept IP signaling streams from an MG, conveying received PRI or CAS trunk signaling. The MGC shall return IP signaling streams to the MG accordingly, for conversion to transmitted PRI or CAS trunk signaling.	5.3.2.12.15	Y	Y	
147	Within the appliance (i.e., LSC or MFSS), the MGC shall use either ITU-T Recommendation H.248 (Gateway Control Protocol Version 3) or a supplier-proprietary protocol to accomplish the MG, trunk, and media stream controls described previously.	5.3.2.12.15	Y	Y	
148	Whenever the MG uses ITU-T Recommendation V.150.1, the following applies: ITU-T Recommendation V.150.1 provides for three states: audio, VBD, and modem relay. After call setup, inband signaling may be used to transition from one state to another. In addition, V.150.1 provides for the transition to FoIP using Fax Relay per ITU-T Recommendation T.38.	5.3.2.12.16	Y	Y	
149	The LSC MG, MFSS MG, and WAN SS MG shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an	5.3.2.12.17	Y	Y	

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
	audible ringing tone) to fail when an internal failure occurs within that MG.				
150	The precedence level and dialed number input to the PEI or AEI shall be as specified in UCR 2008, Section 5.2.3.5.1.2, Interswitch and Intraswitch Dialing.	5.3.2.16	Y	Y	
151	The DSN Worldwide Numbering and Dialing Plan will be used as the addressing schema within the current DSN and its migration into the SIP environment.	5.3.2.16.1	Y	Y	
152	The CCA shall allow session requests from LSC, MFSS EIs, other appliances, and MFSS MGs to contain • Called addresses including DSN numbers from the DSN numbering plan • Called addresses including E.164 numbers from the E.164 numbering plan	5.3.2.16.1	Y	Y	
153	When a session request's called address includes a DSN number from the DSN numbering plan, the CCA shall determine whether the called DSN number is local to the LSC or MFSS, or external to the LSC or MFSS.	5.3.2.16.1	Y	Y	
154	When a session request's called address includes an E.164 number from the E.164 numbering plan, the CCA shall determine whether the called E.164 number is local to the LSC or MFSS, or external to the LSC or MFSS.	5.3.2.16.1	Y	Y	
155	The CCA shall allow each VoIP and Video PEI and AEI served by an LSC or MFSS to have both a DSN number assigned and an E.164 number assigned.	5.3.2.16.1.1	Y	Y	
156	For VoIP and Video PEIs or AEIs that have both a DSN number and an E.164 number assigned, the CCA shall be able to match each PEI's or AEI's DSN number with its E.164 number, and to match each PEI's or AEI's E.164 number with its DSN number.	5.3.2.16.1.1	Y	Y	
157	The CCA shall be able to distinguish DSN called numbers from E.164 called numbers when processing VoIP and Video session requests from PEIs, AEIs, EBCs, MG line cards, and MG trunk groups.	5.3.2.16.1.2	Y	Y	
158	The CCA shall be able to distinguish local [DSN or E.164] called numbers from external [DSN or E.164] called numbers when processing VoIP and Video session requests from PEIs, AEIs, EBCs, MG line cards, and MG trunk groups.	5.3.2.16.1.2	Y	Y	
159	The MFSS or LSC is only required to support one network FQDN for use with SIP URI domain names: "uc.mil" if that appliance is used for SBU traffic, and "cuc.mil" if that appliance is used for classified traffic.	5.3.2.16.1.4.1	Y	Y	
160	The MFSS or LSC is required to ensure that all AS-SIP session requests entering or leaving that appliance use the network FQDN of that appliance (i.e., "uc.mil" for SBU traffic, or "cuc.mil" for Classified traffic) as the domain name in called SIP URIs.	5.3.2.16.1.4.1	Y	Y	
161	All voice systems, TDM or IP technology-based, must contain subscriber assignment information.	5.3.2.16.1.5	Y	Y	
162	Use of the Commercial Cost Avoidance functionality shall be an optional application that can be configured (i.e., enabled and disabled) on each RTS LSC.	5.3.2.23		Y	
163	The LSC shall be able to query the DISN RTS Routing Database on "99 dialed PSTN number" call requests from LSC end users. When the database responds to this query with a DSN number that matches the dialed PSTN number, the LSC shall route the call request over the appropriate IP (AS-SIP) or TDM (e.g., T1.619A PRI) path, using the DSN number returned by the database. When the database responds with a "number not found" indication, the LSC shall route the call request to the local TDM PSTN trunk group (PRI or CAS) on the LSC's MG, using the originally dialed commercial number.	5.3.2.23		Y	
164	The query-response interface between the LSC and the RTS Routing Database shall be LDAP Version 3 (v3) over TLS over IP. This LDAPv3 interface shall be compliant with RFC 4510.	5.3.2.23		Y	
165	The encoding of the LDAPv3 messages and data schema used on the DB query interface between the LSC and the RTS Routing Database shall follow the BER of ASN.1, consistent with Section 5.1, Protocol Encoding, of RFC 4511.	5.3.2.23		Y	

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
166	The DB query interface between the LSC and the RTS Routing Database shall traverse the data firewalls (and not the RTS EBC firewalls) at both the LSC and RTS Routing Database sites.	5.3.2.23		Y	
167	After transmitting a Commercial Cost Avoidance query to the Database, the LSC shall start a “Commercial Cost Avoidance Query Response” timer awaiting a Database response. If the timer expires and no response is received, the LSC shall route the call request to the local TDM PSTN trunk group (PRI or CAS) using the originally dialed commercial number.	5.3.2.23		Y	
168	On Commercial Cost Avoidance call requests that are re-routed to DSN numbers by the database, the LSC shall respond to MFSS or WAN SS AS-SIP signaling indicating that the call was rejected (i.e., an AS-SIP 4xx, 5xx, or 6xx response to an AS-SIP INVITE message), by overflowing these calls from the AS-SIP trunk group to the local TDM PSTN trunk group (PRI or CAS) using the originally dialed commercial number.	5.3.2.23		Y	
169	On Commercial Cost Avoidance call requests that are re-routed to DSN numbers by the database, the MFSS or WAN SS shall accept AS-SIP call requests from the LSC where the DSN number is identified as the called number. The MFSS or WAN SS shall also be capable of returning AS-SIP signaling to the calling LSC that indicates “404 Not Found,” “480 Temporarily Unavailable,” or “500 Server Internal Error.” The MFSS or WAN SS shall be capable of generating this AS-SIP signaling on its own, and shall be capable of relaying that AS-SIP signaling when it is received from a remote MFSS, remote WAN SS, or remote LSC.	5.3.2.23	Y		Y
170	For each RTS end user served by an LSC, the LSC shall be able to upload that user’s DSN phone number, PSTN phone number, and a unique LSC CCA-ID, Primary MFSS/WAN SS CCA-ID, and Backup MFSS/WAN SS CCA-ID to the RTS Routing Database.	5.3.2.23		Y	
171	The AS-SIP signaling appliance shall divert ALL unanswered RTS VoIP calls above the ROUTINE precedence level to a designated RTS DN for PCD (e.g., the number of an attendant console or group of attendant consoles).	5.3.2.25	Y	Y	C
172	Unanswered RTS VoIP calls above the ROUTINE precedence level shall not be forwarded to voicemail, and shall not be forwarded to ACD systems. Instead, they should divert to the PCD DN when the PCD time period expires.	5.3.2.25	Y	Y	C
173	Unanswered RTS VoIP calls at the ROUTINE precedence level shall still be forwarded to voicemail or to ACD systems (when Call Forwarding Don’t Answer is assigned to the called RTS DN), even though PCD is enabled and configured for the AS-SIP signaling appliance.	5.3.2.25	Y	Y	C
174	Calls above the ROUTINE precedence level that are destined to (directly dialed to) DNs assigned to voicemail or ACD systems shall only divert to the PCD DN as specified above (i.e., when they are unanswered at the voicemail or ACD system, and the PCD time period expires).	5.3.2.25	Y	Y	C
175	ROUTINE precedence level calls that are destined to (directly dialed to) DNs assigned to voicemail or ACD systems shall be allowed.	5.3.2.25	Y	Y	C
176	Incoming precedence calls to the attendant’s listed DN, and incoming calls that are diverted to this attendant DN, shall be placed in a queue for the attendant console (or group of attendant consoles).	5.3.2.25	Y	Y	C
177	When a group of attendant consoles on the same LSC is used, and calls are either placed or diverted to the attendant console DN, call distribution across the Console Group shall be used to reduce excessive caller waiting times.	5.3.2.25	Y	Y	C
178	Incoming calls (placed and diverted) to the console DN shall be queued for attendant service by call precedence and time of arrival. The highest precedence call with the longest holding time in the queue shall be offered to an attendant first.	5.3.2.25	Y	Y	C
179	A recorded message of explanation (e.g., ATQA) shall be applied automatically to all the waiting calls in the Attendant Console queue (refer to Table 5.3.4-9, Announcements).	5.3.2.25	Y	Y	C

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
180	The RTS Attendant Console shall interoperate with PBAS/ASAC as described in • Section 5.3.2.7.2.1, PBAS/ASAC Requirements • Section 5.3.2.2.2.3, ASAC – Open Loop • Section 5.3.4.10, Precedence and Preemption The console shall be able to initiate all levels of RTS precedence calls (i.e., ROUTINE through FLASH-OVERRIDE).	5.3.2.26.1	Y	Y	C
181	The RTS Attendant Console shall provide a visual display of each precedence level and the calling number, for incoming direct dialed calls to the attendant, and diverted calls to the attendant (e.g., calls that reach the attendant through PCD).	5.3.2.26.2	Y	Y	C
182	The appliance and the attendant console shall give the attendant the ability to verify and override a busy line condition.	5.3.2.26.4	Y	Y	C
183	If the attendant uses BLV on a called line, and that called line (called EI) is busy, the appliance and the attendant console shall give an audible and visual “called line busy” indication back to the attendant.	5.3.2.26.4	Y	Y	C
184	The appliance and the attendant console shall prevent an attendant from activating BLV or Emergency Interrupt to called lines and called numbers that are located in the commercial network (the PSTN).	5.3.2.26.4	Y	Y	C
185	The appliance and the attendant console shall give the attendant the ability to use Emergency Interrupt to interrupt an existing call on a busy line, and inform the busy user of a new incoming call.	5.3.2.26.4	Y	Y	C
186	The appliance shall give selected destination EIs the ability to be exempt from Emergency Interrupt and attendant break-in.	5.3.2.26.4	Y	Y	C
187	The appliance and the attendant console shall have the ability to route all calls that are normally directed to the console to a separate night service deflection number. The night service deflection number shall be a fixed (preconfigured) or manually-selected DN.	5.3.2.26.5	Y	Y	C
188	When an attendant redirects an incoming call to a destination station, and that station is either busy or does not answer the call within a preset time, the appliance and the attendant console shall ensure that calling party on the redirected call is recalled automatically to the console. In this case, the appliance shall ensure that that the “recalled” call is returned to the console that originally processed the call.	5.3.2.26.6	Y	Y	C
189	The appliance and the attendant console shall have the ability to place calls (both directed to the attendant and diverted to the attendant) into a waiting queue.	5.3.2.26.7	Y	Y	C
190	The appliance and the attendant console shall ensure that calls in the attendant queue are not lost when a console is placed out of service or has its calls forwarded to a night service deflection number.	5.3.2.26.7	Y	Y	C
191	The LSCs that support SIP EIs MUST comply with the differentiated set of requirements defined for SIP EIs if they serve SIP EIs, and LSCs MUST comply with the differentiated set of requirements defined for H.323 and/or vendor-proprietary EIs if they serve H.323 and/or vendor-proprietary EIs.	5.3.4.7.1		Y	
192	All AS-SIP signaling appliances MUST comply with the SIP syntax and encoding rules set forth in RFC 3261. [RFC 3261, Section 25, Augmented BNF for the SIP Protocol]	5.3.4.7.1.1	Y	Y	Y
193	When an AS-SIP signaling appliance does not understand a header field in a request (and support for the header field is not a mandatory requirement under this specification), the AS-SIP signaling appliance MUST ignore that header field and continue processing the message. The AS-SIP signaling appliances MUST ignore any malformed header fields that are not necessary for processing requests.	5.3.4.7.1.3	Y	Y	Y
194	When an AS-SIP signaling appliance, that is implemented as a SIP proxy, receives a SIP Request message, 2xx response, or 18x response, then the AS-SIP signaling appliance MUST add a Record-Route header whereby the userinfo part of the SIP URI is a unique identifier for the AS-SIP signaling appliance and an IP address is used for the host name.	5.3.4.7.1.3c	Y	Y	Y
195	All AS-SIP signaling appliances MUST be call stateful.	5.3.4.7.1.4	Y	Y	Y

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
196	Upon receipt of a new request, AS-SIP signaling appliances MUST perform request validation, route information preprocessing, determine request targets, perform request forwarding, perform response processing, process timer C, handle transport error, handle CANCEL processing, and perform proxy route processing according to RFC 3261	5.3.4.7.1.5	Y	Y	Y
197	All AS-SIP signaling appliances MUST support generation of the long form of the SIP header fields along with the receipt and processing of the long form of the SIP header fields.	5.3.4.7.1.7	Y	Y	Y
198	All AS-SIP signaling appliances MUST support receiving and processing the compact form of the SIP header fields.	5.3.4.7.1.8	Y	Y	Y
199	All AS-SIP signaling appliances serving IP EIs MUST support the offer/answer model for the Session Description Protocol (SDP).	5.3.4.7.1.9	Y	Y	Y
200	If an LSC receives a call request from a served IP EI and the LSC has been unable to establish a TLS connection with its EBC and is unable to do so upon receipt of the INVITE, then the AS-SIP signaling appliance MUST ensure that the IP EI plays the Isolated Code Announcement (ICA) and terminates the call request and MUST send an alarm to the NMS.	5.3.4.7.1.10		Y	
201	When an SS receives an INVITE from either a served LSC or another SS where the Request-URI has a DSN telephone number for which the SS has no entry in its Location Server, then the SS MUST respond with a 404 (Not Found) response code.	5.3.4.7.1.12	Y		Y
202	When an LSC receives an inbound INVITE from its primary (or secondary) SS whose Request-URI has a DSN telephone number for which the LSC has no entry in its Location Server, then the LSC MUST respond with a 404 (Not Found) response message.	5.3.4.7.1.13		Y	
203	The LSCs serving IP EIs MUST ensure that all outbound INVITEs forwarded onto the UC WAN include a Supported header with the option tag “100rel.”	5.3.4.7.1.14		Y	
204	When an AS-SIP signaling appliance receives an INVITE (having an sdp offer) and will be interworking the signaling to the TDM network, then the AS-SIP signaling appliance MUST return an sdp answer in the first non-failure reliable provisional response.	5.3.4.7.1.15	Y	Y	Y
205	When an LSC receives an INVITE (having an sdp offer) intended for a served IP EI, then the AS-SIP signaling appliance MUST NOT return an sdp answer in any provisional response and MUST only place the sdp answer in the 200 response.	5.3.4.7.1.16		Y	
206	When an AS-SIP signaling appliance receives an Empty INVITE (i.e., an INVITE that does not include an sdp offer) and said AS-SIP signaling appliance will be interworking the signaling to the TDM network, then the AS-SIP signaling appliance MUST send an sdp offer in the first reliable non-failure provisional response (1xx response code greater than a 100 response code).	5.3.4.7.1.17	Y	Y	Y
207	When an AS-SIP signaling appliance receives an Empty INVITE intended for a served IP EI, then the AS-SIP signaling appliance MUST NOT send an sdp offer in any provisional response (1xx response code greater than a 100 response code) and MUST only send the sdp offer in the 200 response.	5.3.4.7.1.18	Y	Y	Y
208	When an AS-SIP signaling appliance that is interworking SIP signaling with the TDM network receives a 180 (Ringing) response from the IP network, the AS-SIP signaling appliance MUST ensure that the appropriate ring back tone (e.g., ring back, precedence ring back) is generated on the TDM network.	5.3.4.7.1.19	Y	Y	Y
209	Announcements are not sent in-band on the DSN TDM network; therefore, when an AS-SIP signaling appliance that is interworking SIP signaling with the TDM network receives a 480 (Temporarily Unavailable), 486 (Busy Here), or 488 (Not Acceptable Here) response from the IP network with either no Reason header or a Reason header that does NOT have a preemption cause, the AS-SIP signaling appliance does NOT generate an announcement to be sent to the TDM network, rather it sends either a REL with Q.850 cause code 46 precedence call blocked (in the case of SS7 ISUP) or a Disconnect (in the case of ISDN) with the appropriate cause code message to the TDM network.	5.3.4.7.1.20	Y	Y	Y

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
210	An LSC that receives an outbound call request from a served IP EI MAY include an audio media feature tag and a video media feature tag, as appropriate, in the Contact header field of the INVITE message.	5.3.4.7.1.21		Y	
211	The AS-SIP signaling appliances are NOT required to process and act on the audio media tag and the video media tag in the Contact header but all intermediary AS-SIP signaling appliances MUST preserve the audio media tag (if present) and the video tag (if present) when forwarding the INVITE. (i.e., intermediary AS-SIP signaling appliances MUST NOT strip off or modify the media feature tags).	5.3.4.7.1.22	Y	Y	Y
212	When an LSC receives a call request from a served IP EI intended for a destination outside the enclave, then the AS-SIP signaling appliance MUST generate the P-Asserted-Identity header.	5.3.4.7.1.23		Y	
213	The LSC serving the AS-SIP EI MUST support authentication of the AS-SIP EIs. The user of the AS-SIP EI is required to perform user authentication to the LSC when initiating precedence call requests.	5.3.4.7.1.26		Y	
214	The LSCs serving IP EIs (other than AS-SIP EIs) MUST support authentication of the IP EIs. The user of the IP EI is required to perform user authentication to the LSC when initiating precedence call requests.	5.3.4.7.1.31		Y	
215	When an LSC serving H.323 and/or proprietary EIs receives a request that contains a Require header field with one or more option tags that it does not understand, then it MUST return a 420 (Bad Extension) response code. The response MUST include an Unsupported header field listing those option tags the element did not understand.	5.3.4.7.1.35		Y	
216	The LSCs and AS-SIP EIs MUST support the generating, receiving, and processing of SIP CANCEL requests.	5.3.4.7.2.2		Y	
217	The AS-SIP signaling appliances MUST support the keep-alive mechanism for SIP sessions.	5.3.4.8.1.1	Y	Y	Y
218	The AS-SIP signaling appliances MUST support the generating, receiving, and processing of the Session-Expires and Min-SE header fields.	5.3.4.8.1.3	Y	Y	Y
219	The AS-SIP signaling appliances MUST support the 422 (Session Interval Too Small) response code.	5.3.4.8.1.4	Y	Y	Y
220	The AS-SIP signaling appliances MUST support the option tag “timer” for use with the Supported and Require header fields; however, an AS-SIP signaling appliance acting as a UAC or a SIP EI acting as a UAC MUST NOT place the option tag “timer” in either a Require header or a Proxy-Require header.	5.3.4.8.1.4	Y	Y	Y
221	When an AS-SIP signaling appliance receives an outbound request from a served H.323 or proprietary IP EI, then the AS-SIP signaling appliance MUST operate in accordance with the UAC behavior (when responsible for performing the refresh).	5.3.4.8.1.8	Y	Y	Y
222	When an AS-SIP signaling appliance receives a call request from another AS-SIP signaling appliance, and the destination is a served H.323 or proprietary IP EI, then the AS-SIP signaling appliance MUST operate in accordance with the UAS behavior (when responsible for performing the refresh).	5.3.4.8.1.10	Y	Y	Y
223	When SDP information is present in a SIP message, the SIP message MUST have a content-type header having the MIME Content-Type “application/sdp”.	5.3.4.9.1.2	Y	Y	Y
224	The SDP parser in the AS-SIP signaling appliance (and all AS-SIP EIs, including AS-SIP video conferencing EIs) MUST be able to accept and handle without error any of the SDP line types enumerated in RFC 2327 even if the application ignores the contents.	5.3.4.9.1.3	Y	Y	Y
225	The precedence level of the call request MUST be set forth in a SIP Resource-Priority header field whose syntax is in accordance with RFC 4412, as modified in UCR 2008, Section 5.3.4.10.2	5.3.4.10.2.1	Y	Y	C
226	Video telephony EIs MUST, as the default configuration, require an end user wishing to place a call that includes video, to affirmatively signal the intention to include video to the EI every time the caller wishes to engage in a video telephony call.	5.3.4.12.1.1	Y	Y	C
227	Every time a caller requests a video telephony call and the video portion of the telephony call is successfully established, then the video telephony EI MUST provide the user with an affirmative confirmation	5.3.4.12.1.2	Y	Y	C

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
	that the video is enabled either before, or upon successful completion of, session establishment.				
228	When an INVITE with an sdp offer that includes both audio and video capabilities is received by an LSC serving a destination EI that supports video telephony, then when the call request is received by the destination EI the destination EI MUST indicate to the callee that a telephony call requesting video connectivity has been received.	5.3.4.12.2.1	Y	Y	C
229	Every time a callee accepts a video telephony call and the video portion of the telephony call is successfully established, then the video telephony EI MUST provide the user with an affirmative confirmation that the video is enabled either before or upon successful session establishment.	5.3.4.12.2.3	Y	Y	C
230	AS-SIP Signaling appliances must follow call flows depicted in section 5.3.4.13 for all call features and calling services.	5.3.4.13	Y	Y	Y
231	AS-SIP Signaling appliances must follow requirements depicted in section 5.3.4.14 for all IP to TDM and TDM to IP translations.	5.3.4.14	Y	Y	Y
232	When an interworking AS-SIP signaling appliance receives a request that contains a Require header field with one or more option-tags that it does not understand, then the interworking AS-SIP signaling appliance MUST return a 420 (Bad Extension) response. The response MUST include an Unsupported header field listing those option-tags the element did not understand.	5.3.4.16.1.1	Y	Y	Y
233	All outbound INVITEs generated by an interworking AS-SIP signaling appliance MUST include a Supported header with the option tag “100rel.”	5.3.4.16.1.2	Y	Y	Y
234	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP INVITE requests. Interworking AS-SIP signaling appliances MUST support generating and receiving SIP re-INVITES.	5.3.4.16.2.1	Y	Y	Y
235	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP CANCEL requests.	5.3.4.16.2.2	Y	Y	Y
236	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP OPTIONS requests.	5.3.4.16.2.4	Y	Y	Y
237	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP BYE requests	5.3.4.16.2.5	Y	Y	Y
238	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP ACK requests	5.3.4.16.2.6	Y	Y	Y
239	Interworking AS-SIP signaling appliances MUST support generating and receiving the SIP PRACK method. Interworking AS-SIP signaling appliances MUST support use of the option tag “100rel” with the Require header and Supported header, and MUST support the use of header fields RAck and RSeq.	5.3.4.16.2.8	Y	Y	Y
240	Interworking AS-SIP signaling appliances MUST support generating and receiving the SIP UPDATE method	5.3.4.16.2.9	Y	Y	Y
241	Inter-working AS-SIP signaling appliances MUST be capable of receiving/processing REFER requests, the Refer-To header, and the REFER event package.	5.3.4.16.2.10	Y	Y	Y
242	Interworking AS-SIP signaling appliances MUST support the NOTIFY method for event notification.	5.3.4.16.2.12	Y	Y	Y
243	Interworking AS-SIP signaling appliances MUST, in adherence with the enumerated RFCs, be capable of generating, receiving, and processing the SIP headers listed in UCR 2008 Section 5.3.4.16.3.1	5.3.4.16.3.1	Y	Y	Y
244	The From header MUST include a tag field as specified in RFC 3261, Section 19.3.	5.3.4.16.3.3	Y	Y	Y
245	The To header of a request that is part of a dialog MUST include a tag field as specified in RFC 3261, Section 19.3.	5.3.4.16.3.4	Y	Y	Y
246	Interworking AS-SIP signaling appliances MUST support the use of option tags for the Require, Supported, and Unsupported headers.	5.3.4.16.3.5	Y	Y	Y

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
247	When the interworking LSC sends an initial AS-SIP INVITE to its local EBC intended for its SS, the interworking LSC MUST add two Route header field values, which either takes the form of a route set comprising two Route headers where the first Route header is the sip uri for the EBC at the enclave and the second Route header is the sip uri for the EBC serving the SS, or takes the form of one Route header with two comma-separated field values.	5.3.4.16.3.6		Y	
248	When an interworking SS forwards an initial AS-SIP INVITE to a peer SS, then the interworking SS MUST add a route set comprising two Route headers where the first Route header is the SIP URI for the EBC that serves the interworking SS, and the second Route header is the SIP URI for the EBC serving the peer SS.	5.3.4.16.3.7	Y		Y
249	When an interworking AS-SIP signaling appliance generates an outbound AS-SIP request, the interworking AS-SIP signaling appliance MUST add its own VIA header to the AS-SIP request.	5.3.4.16.3.8	Y	Y	Y
250	When an interworking AS-SIP signaling appliance receives a SIP response to be translated into TDM signaling, then the interworking AS-SIP signaling appliance operates as the UAC for SIP purposes.	5.3.4.16.3.9	Y	Y	Y
251	When an interworking AS-SIP signaling appliance receives an inbound SIP request to be translated into TDM signaling, then the AS-SIP signaling appliance operates as the UAS for SIP purposes.	5.3.4.16.3.10	Y	Y	Y
252	When an interworking AS-SIP signaling appliance generates a SIP response on behalf of a signaling message received from the TDM network, then before forwarding the SIP response the interworking AS-SIP signaling appliance MUST include the VIA headers received in the corresponding SIP request.	5.3.4.16.3.11	Y	Y	Y
253	When an interworking AS-SIP signaling appliance operating as an originating gateway receives an IAM from the TDM network and sends an INVITE to another AS-SIP signaling appliance (SS or LSC), then the interworking AS-SIP signaling appliance MUST add a CCA-ID parameter to the SIP URI of the Contact header populated with its unique identifier before forwarding the INVITE onward to the next AS-SIP signaling appliance.	5.3.4.16.3.12	Y	Y	Y
254	Interworking AS-SIP signaling appliances MUST support generating, receiving, and processing the provisional (1xx) response codes: 100 (Trying), 180 (Ringing), and 183 (Session Progress).	5.3.4.16.4.1	Y	Y	Y
255	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the successful (2xx) response codes: 200 (OK) [RFC 3261, Section 21.2, 200 OK] and 202 (Accepted)	5.3.4.16.4.2	Y	Y	Y
256	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the request failure (4xx) response codes: 400 (Bad Request), 401 (Unauthorized), 403 (Forbidden), 404 (Not Found), 405 (Method Not Allowed), 406 (Not Acceptable), 407 (Proxy Authentication Required), 408 (Request Timeout), 410 (Gone), 413 (Request Entity Too Large), 414 (Request-URI Too Long), 415 (Unsupported Media Type), 416 (Unsupported URI Scheme), 417 (Unknown Resource-Priority), 420 (Bad Extension), 421 (Extension Required), 422 (Session Interval Too Small), 423 (Interval Too Brief), 480 (Temporarily Unavailable), 481 (Call/Transaction Does Not Exist), 482 (Loop Detected), 483 (Too Many Hops), 484 (Address Incomplete), 485 (Ambiguous), 486 (Busy Here), 487 (Request Terminated), 488 (Not Acceptable Here), and 491 (Request Pending).	5.3.4.16.4.4	Y	Y	Y
257	Interworking AS-SIP signaling appliances upon properly receiving a CANCEL request for an INVITE MUST first send a 200 (OK) response code to the CANCEL, and then follow up with a 487 (Request Terminated) response code to the INVITE.	5.3.4.16.4.5	Y	Y	Y
258	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the server failure (5xx) response codes: 500 (Server Internal Error), 501 (Not Implemented), 502 (Bad Gateway), 503 (Service Unavailable), 504 (Server Timeout), 505 (Version Not Supported), 513 (Message Too Large) [RFC 3261, Section 21.5, Server Failure 5xx], and 580 (Precondition Failure)	5.3.4.16.4.6	Y	Y	Y
259	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the global failures (6xx) response codes: 600 (Busy Everywhere), 603 (Decline), 604 (Does Not Exist Anywhere), and 606 (Not Acceptable).	5.3.4.16.4.7	Y	Y	Y

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
260	When an interworking AS-SIP signaling appliance receives an outbound request from the PSTN (i.e., the interworking AS-SIP signaling appliance is operating as an originating gateway) and the destination is NOT an IP EI directly served by the interworking AS-SIP signaling appliance, then the interworking AS-SIP signaling appliance MUST operate in accordance with the UAC behavior set forth in RFC 4028.	5.3.4.17.1.1	Y	Y	Y
261	When an interworking AS-SIP signaling appliance acting as a terminating gateway receives a call request from another AS-SIP signaling appliance, then the interworking AS-SIP signaling appliance MUST operate in accordance with the UAS behavior set forth in RFC 4028.	5.3.4.17.1.3	Y	Y	Y
262	Interworking AS-SIP signaling appliances MUST support the option tag “resource-priority” for use with the Require header.	5.3.4.18.3.1	Y	Y	Y
263	The interworking AS-SIP signaling appliance MUST receive and accept a Require header field with the option tag “resource-priority” in the INVITE, UPDATE, and REFER messages. Interworking AS-SIP signaling appliances MUST NOT reject the message with a 420 (Bad Extension) response code, but rather it MUST accept the request and translate it into the appropriate TDM signaling message as required.	5.3.4.18.3.2	Y	Y	Y
264	If an interworking AS-SIP signaling appliance receives an inbound ROUTINE call request over the IP network for a destination on the TDM network and the interworking AS-SIP signaling appliance has insufficient bandwidth-related resources (e.g., lack of circuit-switched trunk capacity for bearer traffic) to handle the call request, the interworking AS-SIP signaling appliance MUST reply with a 488 (Not Acceptable Here) response code and SHOULD include a Warning header with warning code 370 (Insufficient Bandwidth).	5.3.4.18.4.5	Y	Y	Y
265	If an interworking AS-SIP signaling appliance receives an inbound precedence call request (i.e., with precedence level PRIORITY or above) over the IP network for a destination on the TDM network and the interworking AS-SIP signaling appliance has insufficient bandwidth-related resources (e.g., lack of circuit-switched trunk capacity for bearer traffic) to handle the call request, and if there are insufficient existing calls (and/or call requests) of lower precedence whose removal would provide the necessary resources to support the pending call request, then: - The interworking AS-SIP signaling appliance MUST reply with a 488 (Not Acceptable Here) response code and SHOULD include a Warning header with warning code 370 (Insufficient Bandwidth), and - The AS-SIP signaling appliance serving the calling IP EI MUST arrange for a BPA to be played to the calling IP EI before terminating the call.	5.3.4.18.4.6	Y	Y	Y
266	When an interworking AS-SIP signaling appliance receives a precedence call request from the IP network that it translates and forwards onto the TDM network and the response from the TDM network is a REL with Q.850 cause code 46 precedence call blocked (in the case of SS7 ISUP) or a Disconnect with Q.850 cause code 46 precedence call blocked (in the case of ISDN), then the interworking AS-SIP signaling appliance MUST generate a 488 (Not Acceptable Here) response that SHOULD include a “Warning” header with warning code 370 (Insufficient Bandwidth) with no Reason header that it sends onto the IP network.	5.3.4.18.6.2	Y	Y	Y
267	Interworking AS-SIP signaling appliances directly serving IP EIs MUST comply with the requirements in UCR 2008, Section 5.3.4.10.3.3.2, Implementing the Network Preemption. In addition, interworking AS-SIP signaling appliances directly serving IP EIs MUST meet the enumerated requirements in section 5.3.4.18.6.3.2.	5.3.4.18.6.3	Y	Y	Y
268	AS-SIP signaling appliances must follow all call flows depicted in UCR 2008 Section 5.3.4.19 for all supplementary services.	5.3.4.19	Y	Y	Y
269	The product shall support dual IPv4 and IPv6 stacks as described in RFC 4213.	5.3.5.4	Y		Y
270	Dual stack end points or Call Control Agents shall be configured to choose IPv4 over IPv6.	5.3.5.4	Y		Y

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
271	All nodes that are "IPv6-capable" shall be carefully configured and verified that the IPv6 stack is disabled until it is deliberately enabled as part of a risk management strategy.	5.3.5.4	Y		Y
272	The product shall support the IPv6 format as described in RFC 2460 and updated by RFC 5095.	5.3.5.4	Y		Y
273	The product shall support the transmission of IPv6 packets over Ethernet networks using the frame format defined in RFC 2464.	5.3.5.4	Y		Y
274	The product shall support a minimum MTU of 1280 bytes.	5.3.5.4.1	Y		Y
275	The product shall not use the Flow Label field as described in RFC 2460. The product shall be capable of setting the Flow Label field to zero when originating a packet. The product shall not modify the Flow Label field when forwarding packets. The product shall be capable of ignoring the Flow Label field when receiving packets.	5.3.5.4.2	Y		Y
276	The product shall support the IPv6 Addressing Architecture as described in RFC 4291.	5.3.5.4.3	Y		Y
277	The product shall support the IPv6 Scoped Address Architecture as described in RFC 4007.	5.3.5.4.3	Y		Y
278	The product shall support Neighbor Discovery for IPv6 as described in RFC 2461 and RFC 4861.	5.3.5.4.5	Y		Y
279	The product shall not set the override flag bit in the Neighbor Advertisement message for solicited advertisements for anycast addresses or solicited proxy advertisements.	5.3.5.4.5	Y		Y
280	When a valid "Neighbor Advertisement" message is received by the product and the product neighbor cache does not contain the target's entry, the advertisement shall be silently discarded.	5.3.5.4.5	Y		Y
281	When a valid "Neighbor Advertisement" message is received by the product and the product neighbor cache entry is in the INCOMPLETE state when the advertisement is received and the link layer has addresses and no target link-layer option is included, the product shall silently discard the received advertisement.	5.3.5.4.5	Y		Y
282	When address resolution fails on a neighboring address, the entry shall be deleted from the product's neighbor cache.	5.3.5.4.5	Y		Y
283	The product shall support the ability to configure the product to ignore Redirect messages. The product shall only accept Redirect messages from the same router as is currently being used for that destination.	5.3.5.4.5.1	Y		Y
284	If the product supports routing functions, the product shall inspect valid router advertisements sent by other routers and verify that the routers are advertising consistent information on a link and shall log any inconsistent router advertisements. The product shall prefer routers that are reachable over routers whose reachability is suspect or unknown.	5.3.5.4.5.2	Y		Y
285	The product shall support manual assignment of IPv6 addresses.	5.3.5.4.6	Y		Y
286	The product shall support the ICMPv6 as described in RFC 4443. The product shall have a configurable rate limiting parameter for rate limiting the forwarding of ICMP messages.	5.3.5.4.7	Y		Y
287	The product shall support the capability to enable or disable the ability of the product to generate a Destination Unreachable message in response to a packet that cannot be delivered to its destination for reasons other than congestion.	5.3.5.4.7	Y		Y
288	The product shall support the enabling or disabling of the ability to send an Echo Reply message in response to an Echo Request message sent to an IPv6 multicast or anycast address.	5.3.5.4.7	Y		Y
289	The product shall validate ICMPv6 messages, using the information contained in the payload, before acting on them.	5.3.5.4.7	Y		Y
290	The product shall support MLD as described in RFC 2710.	5.3.5.4.8	Y		Y

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
291	For traffic engineering purposes, the bandwidth required per voice subscriber is calculated to be 110.0 kbps (each direction) for each IPv6 call.	5.3.5.4.11	Y		Y
292	The product shall forward packets using the same IP Version as the Version in the received packet.	5.3.5.4.12	Y		Y
293	The product shall use the Alternative Network Address Types (ANAT) semantics for the Session Description Protocol (SDP) in accordance with RFC 4091 when establishing media streams from dual-stacked appliances for AS-SIP signaled sessions.	5.3.5.4.12	Y		Y
294	The product shall prefer any IPv4 address to any IPv6 address when using ANAT semantics.	5.3.5.4.12	Y		Y
295	The product shall place the option tag "SDP-ANAT" in a Required header field when using ANAT semantics in accordance with RFC 4092.	5.3.5.4.12	Y		Y
296	The products shall support Differentiated Services as described in RFC 2474 for a voice and video stream in accordance with Section 5.3.2, Assured Services Requirements, and Section 5.3.3, Network Infrastructure E2E Performance Requirements, plain text DSCP plan.	5.3.5.4.14	Y		Y
297	The LSC must meet all requirements for FCAPS Management and audit logs as listed in UCR 2008 section 5.3.2.7.2.6	5.3.2.7.2.6		Y	
298	The physical interface between the DISA VVoIP EMS and the network components (i.e., LSC, MFSS, EBC, CE Router) is a 10/100-Mbps Ethernet interface. The interface will work in either of the two following modes using auto-negotiation: IEEE, Ethernet Standard 802.3, 1993; or IEEE, Fast Ethernet Standard 802.3u, 1995.	5.3.2.4.4	Y	Y	Y
299	Redundant physical Ethernet interfaces are required for signaling and bearer traffic. If the primary signaling and bearer Ethernet interface fails, then traffic shall be switched to the backup signaling and bearer Ethernet interface.	5.3.2.4.4	Y	Y	Y
300	As specified in Section 5.3.2.4.4, VoIP NMS Interface Requirements, the MFSS, WAN SS, and LSC components shall support at least one pair of physical Ethernet management interfaces at the component level (not at the device level). One of these Ethernet management interfaces shall be used for component-level communication with a Local EMS. The other Ethernet management interface shall be used for component-level communication with the remote VVoIP EMS.	5.3.2.17.2	Y	Y	Y
301	A network appliance shall have Operations interfaces that provide a standard means by which management systems can directly or indirectly communicate with and, thus, manage the various network appliances in the DISN.	5.3.2.17.2	Y	Y	Y
302	There shall be a local craftsperson interface (Craft Input Device (CID)) for OA&M for all VVoIP network components.	5.3.2.17.2	Y	Y	Y
303	The network appliances shall provide NM data to the external VVoIP EMS.	5.3.2.17.2	Y	Y	Y
304	A network appliance shall communicate with an external Voice and Video management system by a well-defined, standards-based management interface using an industry-accepted management protocol.	5.3.2.17.2	Y	Y	Y
305	Communications between VVoIP EMS and the VVoIP network appliances shall be via IP.	5.3.2.17.2	Y	Y	Y
306	A network appliance shall issue state change notifications for changes in the states of replaceable components, including changes in operational state or service status, and detection of new components.	5.3.2.17.2	Y	Y	Y
307	A network appliance shall be provisioned by the VVoIP EMS with the address, software, and OSI Layer 4 port information associated with its Core Network interfaces.	5.3.2.17.2	Y	Y	Y
308	A network appliance shall be capable of maintaining and responding to VVoIP EMS requests for resource inventory, configuration, and status information concerning Core Network interface resources (e.g., IP or MAC addresses) that have been installed and placed into service.	5.3.2.17.2	Y	Y	Y

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
309	Network appliances that provide voice and video call service shall have the capability to invoke traffic flow (NM) controls as detailed in Section 5.3.2.18, Network Management Requirements of Appliance Functions.	5.3.2.17.2	Y	Y	Y
310	A network appliance shall be capable of setting the Administrative state and maintaining the Operational state of each Core Network interface, and maintaining the time of the last state change.	5.3.2.17.2	Y	Y	Y
311	Alarm messages must be distinguishable from administrative log messages.	5.3.2.17.3.1.1	Y	Y	Y
312	The NEs shall detect their own fault (alarm) conditions.	5.3.2.17.3.1.2	Y	Y	Y
313	The NEs shall generate alarm notifications.	5.3.2.17.3.1.3	Y	Y	Y
314	The network elements shall send the alarm messages in NRT. More than 99.95 percent of alarms shall be detected and reported in NRT. Near Real Time is defined as event detection and alarm reporting within 5 seconds of the event, excluding transport time.	5.3.2.17.3.1.4	Y	Y	Y
315	The network components shall send alarm messages in SNMPv3 format.	5.3.2.17.3.1.5	Y	Y	Y
316	Capability to access and modify configuration data by the VVoIP EMS shall be controllable by using an access privileges function within the network appliance.	5.3.2.17.3.2.1	Y	Y	Y
317	The VVoIP NEs shall be able to receive and respond to remote NM commands.	5.3.2.17.3.4.2	Y	Y	Y
318	When ASAC budgets are reduced, by NM action, below the current budget allocation, any previous sessions (regardless of precedence level) in excess of the new budget shall be allowed to terminate naturally. This assumes that the CE Router queue bandwidths would not be reduced until the LSC session count fell below or equal to the newly commanded reduced budget, to prevent the corruption of existing sessions.	5.3.2.17.3.4.2.2	Y	Y	Y
319	The LSC, MFSS, and WAN SS shall have the capability of setting the percentage of calls to be blocked to the designated destination(s).	5.3.2.17.3.4.2.7	Y	Y	Y
320	FLASH and FLASH-OVERRIDE calls shall not be affected by NM controls.	5.3.2.17.3.4.2.7	Y	Y	Y
321	Within IP, directionalization is controlled by designating all or part of the call budget as inbound (i.e., local destination) and/or outbound (i.e., local origination). The default is no designation (i.e., calls up to the total budget can be inbound or outbound in any combination). It does not change the total budget, only the sourcing direction of the budget; therefore, there is no impact to the router queue bandwidths.	5.3.2.17.3.4.2.10	Y	Y	Y
322	Within IP, the routing of all traffic (i.e., VVoIP and non-VVoIP) is handled via MPLS in the DISN core. The MPLS automatically finds the most effective route for the traffic.	5.3.2.17.3.4.2.11	Y	Y	Y
323	The WAN-level ASAC must be able to account for each subtended LSC under its control. Therefore, the MFSS and WAN SS ASAC must be able to set call budgets for multiple LSC locations via the VVoIP EMS and local EMS access points.	5.3.2.17.3.4.2.13	Y		Y
324	The LSC-level ASAC is required to only account for itself. Therefore, the LSC ASAC must be able to set call budgets for only the PEI/AEIs under its control via the VVoIP EMS and local EMS access points.	5.3.2.17.3.4.2.13		Y	
325	The product shall have the capability of setting a PEI/AEI's maximum allowed precedence level for originating a call. This is a "subscriber class mark feature," which is controlled by the LSC system administrator.	5.3.2.17.3.4.2.14		Y	
326	The product shall have the capability of controlling the destination(s) that a PEI or AEI is restricted from calling. This is a subscriber class mark feature that is controlled by the LSC system administrator.	5.3.2.17.3.4.2.14		Y	
327	The ASAC must provide the separate counts for voice and video, in 5-minute intervals. The MFSS and WAN SS ASAC must provide these counts for each of the subtended LSCs under its control, while the LSC is only to provide these counts for the PEIs/AEIs that it controls.	5.3.2.18.2	Y	Y	Y

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
328	A switching network appliance shall acquire, activate, and manage a CCA software download as directed by the Local EMS. The CCA software may be managed on a per CCA hardware component basis.	5.3.2.18.3.1.1	Y	Y	Y
329	The CCA shall be able to manage the following parameters in the CCA from the VVoIP EMS: • CCA Identification parameter • Recording Office Identification parameter	5.3.2.18.3.1.1	Y	Y	Y
330	The CCA shall manage the activation and deactivation of service features. The CCA shall maintain data for the media server and UFS functions it interacts with. The CCA shall be able to create a backup and manage restoration of configuration data by placing its stable data and changes to the latest configuration in a nonvolatile storage device.	5.3.2.18.3.2	Y	Y	Y
331	A CCA shall meet all applicable Operations Technology Generic Requirements (OTGR) for switching system NE trouble isolation in Telcordia Technologies GR-474-CORE. A CCA shall perform root-cause analysis for any faults within its purview, report the root cause, and suppress the reporting of non-root-cause conditions. A CCA shall support the ability to perform internal diagnostics on its call processing functionality and internal resources, initiated either locally or upon request by the VVoIP EMS.	5.3.2.18.3.3	Y	Y	Y
332	The CCA shall provide trunk group-related traffic measurements as specified in Telcordia Technologies GR-477-CORE, Section 4.1.3. For all calls originating at a CCA, the CCA shall monitor call set-up delay statistics, including delay incurred as part of the set-up of the core network bearer connection.	5.3.2.18.3.5	Y	Y	Y
333	An MG shall manage logical and physical resource inventory information. An MG shall issue an autonomous notification to the VVoIP EMS whenever a new inventory or capabilities are added, or configuration is changed through local management activity. An MG maintains the information related to service features and data, including the management of service logic.	5.3.2.18.5.1	Y	Y	Y
334	An MG shall manage current MG state and status information about its installed major components, line and plug-in cards, and processes.	5.3.2.18.5.1.2	Y	Y	Y
335	Upon the detection or clearing of alarm conditions, the MG shall generate and forward, based on filtering criteria, a notification to the VVoIP EMS. An MG shall support queries for alarm status, state, and current problem information. An MG shall monitor, detect, and generate alarm conditions and states associated with hardware, functional components, system interfaces, and logical resources (e.g., trunk terminations, tone and announcement generators, media content detectors, signal processors, echo control devices).	5.3.2.18.5.2.1	Y	Y	Y
336	An MG shall perform root-cause analysis for any faults within its purview, report the root cause, and suppress the reporting of non-root-cause conditions.	5.3.2.18.5.2.2	Y	Y	Y
337	An MG shall, on request or per a pre-established schedule, run diagnostics on internal resources, hardware, or software, and report the result to the VVoIP EMS.	5.3.2.18.5.2.3	Y	Y	Y
338	An MG shall provide both local and remote loopback capabilities for the digital interfaces that terminate at the MG ports.	5.3.2.18.5.2.3	Y	Y	Y
339	Upon receiving a request from the VVoIP EMS or by an established schedule, an MG shall provide a report of a parameter's present or history counters.	5.3.2.18.5.3.1	Y	Y	Y
340	An MG shall generate TCAs to notify the VVoIP EMS when a thresholded count exceeds its threshold during a measurement interval.	5.3.2.18.5.3.2	Y	Y	Y
341	The MG shall manage interexchange trunk (between MG and SSP), trunk group, trunk, and physical resource inventory and configuration data. The MG shall manage MG termination-related status information.	5.3.2.18.5.4	Y	Y	Y
342	An MG shall, on request or on schedule, run diagnostics on internal resources and hardware, run checks on software, and report the results to the VVoIP EMS. The MG shall provide test access to external test equipment for passively monitoring the traffic through the MG interfaces. This passive monitoring shall not degrade the performance of traffic.	5.3.2.18.5.5	Y	Y	Y

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

ID	Requirement	UCR Ref (UCR 2008 CH1)	MFSS	LSC	WAN SS
343	The MG shall receive voice-grade analog line configuration data from the VVoIP EMS upon service activation.	5.3.2.18.5.7	Y	Y	Y
344	The MG shall provide diagnostic tests to detect and verify faults, such as low loop resistance or ground conditions, or any other faults within the MG that could cause false ring trip or false answer.	5.3.2.18.5.8	Y	Y	Y
345	The MG shall support the collection of the standard DS1, DS3, Physical Layer Convergence Protocol (PLCP), SONET, and ISDN BRI line performance monitoring requirements, as defined in Telcordia Technologies GR-820-CORE, for applicable interfaces.	5.3.2.18.5.9	Y	Y	Y
346	For the selected recording format that is chosen, of all the call information that will be provided, the following call data shall be provided in the record data: 1. Host Name of the CCA controlling the call processing. 2. Start Date of call (In Julian or Calendar). 3. Start Time of Call (Hour + Minute + Second). 4. Elapsed Time of Call and/or Stop Time of call. 5. Calling Number. 6. Called Number (included all dialed digits).	5.3.2.19.2.1	Y	Y	Y
347	For the selected recording format that is chosen, of all the call information that will be provided, the following call data shall be provided in the record data if it applies to the call: Conference Call Indicator.	5.3.2.19.2.1	Y	Y	Y
348	The product shall provide a voice quality record at the completion of each voice session. The voice quality record shall be included in the CDR that the LSC, MFSS, or WAN SS generates for that session, and shall conform to the E-Model, as described in TIA TSB-116-A, and ITU-T Recommendation G.107. The voice quality record shall contain the calculated R-Factor for the Voice session per TIA TSB-116-A.	5.3.2.19.2.1.1	Y	Y	Y
349	As part of the voice quality record, the product shall provide the raw voice session statistics that are used to make the R-Factor calculation to include, as a minimum, the latency, packet loss, Equipment Impairment Factor (Ie), and the TCLw. The product shall provide the jitter for the session.	5.3.2.19.2.1.1	Y	Y	Y
350	The product shall generate an alarm to the VVoIP EMS when the session R-Factor calculation in the CDR fails to meet a configurable threshold. By default, the threshold shall be an R-Factor value of 80, which is equivalent to an MOS value of 4.0.	5.3.2.19.2.1.1	Y	Y	Y
351	The mass storage in the BA must be non-volatile. The mass storage in the BA must be able to retain at least five average-busy-season business days of AMA data. (NOTE: This is needed to provide adequate capacity for high-volume storage of CDRs.)	5.3.2.19.2.3	Y	Y	Y
352	The BA should be able to output the records electronically over a secured connection. The BA should have the ability to transfer the records to a physical storage media that is also removable.	5.3.2.19.2.4	Y	Y	Y

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

LEGEND:

ACD	Automatic Call Distributor	LAN	Local Area Network
AEI	AS-SIP End Instrument	LDAP	Lightweight Directory Access Protocol
AMA	Automatic Message Accounting	LDAPv3	Lightweight Directory Access Protocol version 3
ANAT	Alternative Network Address Types	LSC	Local Session Controller
ANSI	American National Standards Institute	MAC	Media Access Control
ASAC	Assured Services Admission Control	Mbps	Megabits per second
ASLAN	Assured Services Local Area Network	MFS	Multifunction Switch
AS-SIP	Assured Services Session Initiation Protocol	MFSS	Multifunction SoftSwitch
ATQA	Attendant Queue Announcement	MG	Media Gateway
B2BUA	Back-to-back User Agent	MGC	Media Gateway Controller
BA	Billing Agent	MLD	Multicast Listener Discovery
BLV	Busy Line Verification	MLPP	Multilevel Precedence and Preemption
BNF	Backus-Naur Form	Modem	Modulator/Demodulator
C	Conditional	MoIP	Modem over Internet Protocol
C2	Command and Control	MOS	Mean Opinion Score
CAC	Common Access Card	MPLS	Multiprotocol Label Switching
CAS	Channel Associated Signaling	ms	milliseconds
CCA	Call Control Agent	MTU	Maximum Transmission Unit
CCS7	Common Channel Signaling 7	NE	Network Element
CDR	Call Data Record	NM	Network Management
CE	Customer Edge	NMS	Network Management System
CF	Call Forward	OA&M	Operations, Administration, and Maintenance
CH1	Change 1	OCONUS	Outside the Continental United States
CID	Craft Input Device	OSI	Open Systems Interconnect
CND	Calling Number Delivery	OTGR	Operations Technology Generic Requirements
CONUS	Continental United States	PBAS	Precedence Based Assured Services
D-Channel	Data Channel	PBX	Private Branch Exchange
DB	Database	PCD	Precedence Call Diversion
DHCP	Dynamic Host Configuration Protocol	PCM	Pulse Code Modulation
DISA	Defense Information Systems Agency	PEI	Proprietary End Instrument
DISN	Defense Information System Network	PLCP	Physical Layer Convergence Protocol

Table 3-1. LSC Products Capability/Functional Requirements Table (continued)

LEGEND (Continued):

DN	Directory Number
DoD	Department of Defense
DS1	Digital Signal Level 1
DS3	Digital Signal Level 3
DSCP	Differentiated Services Code Point
DSN	Defense Switched Network
E2E	End-to-end
EBC	Edge Boundary Controller
EC	Echo Canceller
EI	End Instrument
EMS	Element Management System
EO	End Office
ETSI	European Telecommunications Standards Institute
FCAPS	Fault, Configuration, Accounting, Performance, and Security
FoIP	Fax over Internet Protocol
FQDN	Fully Qualified Domain Name
G3 Fax	Group 3 Facsimile
Hz	Hertz
IAD	Integrated Access Device
IAW	In Accordance With
ICA	Isolated Code Announcement
ID	Identification
ICMPv6	Internet Control Message Protocol for IPv6
Ie	Equipment Impairment Factor
IEEE	Institute of Electrical and Electronics Engineers, Inc.
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPSec	Internet Protocol Security
IPv4	Internet Protocol Version 4
IPv6	Internet Protocol Version 6
ISDN	Integrated Services Digital Network
ISUP	ISN User Part
ITU-T	International Telecommunications Union – Telecommunication Standardization Sector
IWF	Interworking Function
kbps	Kilobits per second
PRI	Primary Rate Interface
PSTN	Public Switch Telephone Network
REL	Release Message
RFC	Request For Communication
RTS	Real Time Services
SAC	Session Admission Control
SBU	Sensitive, but Unclassified
SCIP	Secure Communications Interoperability Protocol
SCS	Session Control and Signaling
SDP	Session Description Protocol
SG	Signaling Gateway
SIP	Session Initiation Protocol
SMEO	Small End Office
SONET	Synchronous Optical Network
SRTCP	Secure Real-Time Transport Control Protocol
SRTP	Secure Real-Time Transport Protocol
SS	Softswitch
SS7	Signaling System number 7
SUT	System Under Test
TA	Terminal Adaptor
TCA	Threshold Crossing Alert
TCLw	Weighted Terminal Coupling Loss
TDM	Time Division Multiplexing
TIA	Telecommunications Industry Association
TLS	Transport Layer Security
UAC	User Agent Client
UAS	User Agent Server
UC	Unified Capabilities
UCR	Unified Capabilities Requirements
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
U.S.	United States
VBD	Voice Band Data
VoIP	Voice over Internet Protocol
VVoIP	Voice and Video over Internet Protocol
WAN	Wide Area Network
Y	Yes